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PART I: Selected Papers from KEAMSAC2015

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Integrating Composer/Improviser's Actions in the creation of an interactive and provocative composition system

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This paper discusses computer aided and automated composition/analysis, and related aesthetic understanding. It discusses the program *ICIA*, prototyped with the Max environment, that integrates: sound analysis systems; sound effecting and manipulation systems; algorithmic composition processes; and gesture recognition systems.

ICIA segments sounds selected by the composer/improviser into usable parts and groupings via a user and machine defined stochastic processes. The groupings are represented on a display for manipulation through a variety of compositional processes by the composer/improviser in real time. The sounds that the composer/improviser may use include: the sounds of the environment; sounds previously stored; sounds being generated by *ICIA*; or any other kind of audible input the composer/improviser chooses.

It provides a provocative environment for the composer/improviser by interpreting their physical, sonic, intentional and non-intentional actions and decisions through a set of algorithms that the composer/improviser has varying control over. These algorithms have built into them a 'degree of predictability' scale that the composer or machine may set. Terms such as 'more' or 'less', 'inside' or 'outside', 'regular or 'irregular' are also used in relation to aspects of the compositional/improvisational process. Using such 'non-musical' terms causes the composer/improviser to consider their approach outside of traditional composition systems. Therefore the composer/improviser is provoked to take into account all aspects of their interaction with *ICIA* in the creation of a sound art/musical artwork, exposing their conscious and unconscious, intentional and non-intentional, aesthetic approaches in the process of creation.

Curtis Roads wrote "As I listen and compose, I inevitably formulate new techniques and aesthetic concepts" (2015). The formulation of new techniques, often based on the development of new technologies, is continuous in most walks of life; the arts is where these can develop into new aesthetics.

The development of computer based algorithmic composition and improvisation has a comparatively short but dense history, ranging through: processes (Edwards 2011; Alpern 1995; Nierhaus 2009; Fernández / Vico 2013; Simoni / Dannenberg 2013; Mazurowski 2015), considerations (Manzolli et al. 1999; Harley 1995; Edwards 2011; Alsop 1999; Supper 2001; Jacob 1996; Williams et al. 2014; Chagas 2014; Roads 2015), and histories (Ariza 2005; Doornbusch 2002; Ariza 2011), and this list is at best an indicative tip of the iceberg.

Any artist of any modality could make Roads's quote; it is a function of creativity. Often the new techniques and aesthetic concepts are formulated and expressed within well-delineated aesthetics, and these are for the most part, based in pre-existing techniques and aesthetic concepts.

An example of this can be seen in Picasso's development of figurative painting, where the images in the 'Blue period', through the Rose and African periods, to Cubism in which contemporary Georges Braque was also a seminal motivator.

The same can be seen in the C20 development of 'serial' music. From the tonal manipulations of Wagner, Schoenberg 'emancipated the dissonance', taught composition to Webern, Berg, Harrison, and Cage and others, who then developed Schoenberg's aesthetic and technical approaches in a variety of ways, to idiomatically form systematic/algorithmic approaches to musical composition in the C20 that are expressive and idiosyncratic to each composer and their individual aesthetic(s).

Here the development of new artworks came about in pictorial art and sonic art through almost universally systems: the tempered pitch system of western music and the representative approach of painting. In both cases the functional aspects of each art form were explored and interrogated, resulting in new processes and nuanced outcomes, offering new interpretation of the traditions with the art forms.

Overview of ICIA

Integrating Composer Improviser Actions (ICIA) is designed to provoke this situation by taking the initial ideas of the composer – potentially expressed as: motif or gesture deliberately constructed or chosen for development, a structure to be populated and explored, a recorded improvisation (be that in memory, notation, or as a sonic object), a sonic representation of a non musical event or object, and so on; and all of these approaches may overlap and intertwine.

It is made up of a number of simple algorithmic processes that when concatenated create a vast variety of possible outcomes.

Algorithmic processes

ICIA sees the three main attributes music as essentially events that happen in the domains of: pitch, time, and amplitude; these attributes are what is performed with a specific timbre or timbres, and the relationships between these events are what is experienced by the listener. Composition is considered as arranging relationship alterations between intervals. These can be intervals that include the attributes:

- pitch/frequency,
- time (duration and onset),
- amplitude,
- density/polyphony,
- timbre,
- predictability,
- structure,

and these can be considered as: relationships between events/attributes, as set points, and as a mix of the two. By arranging attribute relationships: for example the differences between frequencies experienced in the sound domain and in the temporal domain, a composition is formed.

For ease ICIA initially sees frequencies expressed as MIDI notes, and the intervals between them are seen as ratio based relationships. Figure 1 shows a relational nine-note sequence where the intervals from the original note (C60) are set. ICIA considers notes differently to pitch classes, for example the note F#4 (MIDI note 66) is different to the note F#3 (MIDI note 54).

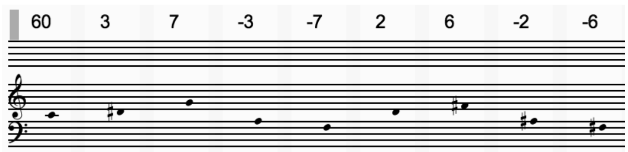

Figure 1. Note interval relationships to a fixed note point

Figure 2 shows numeric relationships to a fixed value (222), which is multiplied by the numbers in the top row.

222.	1.	2.	3.	1.	2.	3.5	2.5	4.	4.5
222.	222.	444.	666.	222.	444.	777.	555.	888.	999.

Figure 2: Numeric interval relationships to a fixed point

The values shown in Figure 2 can be used to influence many parameters, such as structure, tuning, amplitude, timbre, temporal relationships and so on. This information can be routed to a variety of outcomes via MatrixMapper, shown in Figure 9, through which any attribute can be affected by simple arithmetic operations and sent to a variety of inputs affecting their behaviour.

These operations are simply multiplying the input number, filtering it though a modulus, and adding a positive or negative number to it. This process can have profound effect, for example, the note string in Figure 1 was passed through the operators multiply (*) by 23, filter through the modulo (%) 48, the add (+) 48: the resulting note string is shown in reverse in Figure 3: Resulting note string from operations 1, and Figure 4: Resulting note string from operations 2, shows the result when the string in Figure 1 was passed through the arithmetic operators * 13, % 48, + 48.

These notes are in reverse as that is the order in which they would be heard and the operations would occur, that is, the first note C60 is transformed in to note C84, then note D#63 is transformed into note A57, and so on.

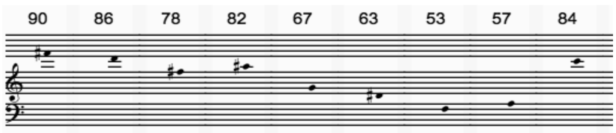

Figure 3: Resulting note string from operations 1

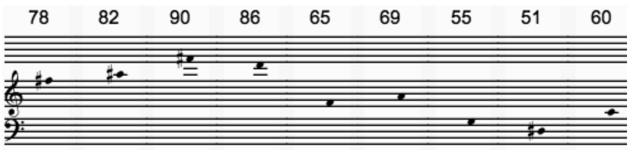

Figure 4: Resulting note string from operations 2

These note strings can also be represented as frequency strings, for example the frequencies seen in Figure 1 can be seen in Figure 5.

184.99721	233.08188	369.99442	293.66477	
174.61412	220.	391.99544	311.12698	261.62557

Figure 5: Frequencies of notes in Figure 1 (shown in reverse order)

Figure 6 shows the flow of data in ICIA. Starting from the Generator, which may contain information that is input by the composer or improviser, a matrix, shown in Figure 9 and Figure 10, that maps the data across 8 possible outputs. These outputs are the effected and returned to a subsequent matrix before the sonic rendering.

The frequencies shown in Figure 5 (or any other data generated) may be reinterpreted into values that relate to: durations, amplitudes, density-of-events, timbre selection or modification, predictability of change and/or overall structure. An example of this reinterpretation could be using the frequencies to generate the amplitudes, durations, and onset times of the nine-note sequence. For example: MIDI note C4 could be assigned the duration of 262 msecs (rounded up frequency value) an amplitude of 0.311 (the frequency of the adjacent note), an onset time of the next sound of 391 msecs, two note polyphony (220 rounded down) of subsequent frequencies 174.61412 and 293.66477, the synthesis of the sound has wave form amplitudes of 0.293 (saw wave), 0.396 (triangle wave), and 0. 233 (rectangle wave), and the depth of reverberation for the concatenating heard event is set 0.184.

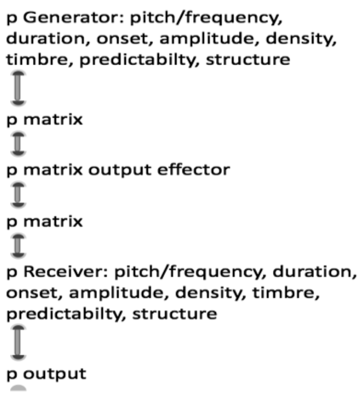

Figure 6: Flow of data in ICIA

Compositional input

Two approaches can be used by the composer/improviser to input to ICIA, NumInput, shown in Figure 7 and NoteInput, shown in Figure 8.

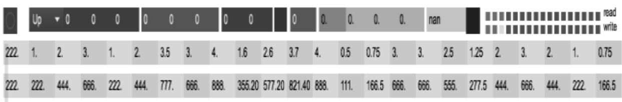

Figure 7: NumInput – numeric data input to ICIA

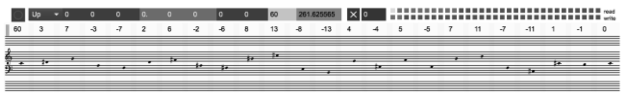

Figure 8: NoteInput – MIDI bases note input to ICIA

Here the data put into the object shown in Figure 7 can relate to any variable that may be used in the creation of the composition, and the note values shown in Figure 8 can be used to represent either numbers or frequencies, which can then be mapped to any other output, as shown in Figure 9 and Figure 10.

MatrixMapper

The MatrixMapper, shown in Figure 9 and Figure 10, is used to receive input from NumericInput and NoteInput, shown in Figure 7 and Figure 8, effect and send data to selected outputs. Figure 9: MatrixMapper - Direct map from input to output after effector, shows eight elements; here listed as pitch, freq., duration, etc. Here the input data of each element is mapped to a corresponding element after effected in the effector segment. In Figure 10: MatrixMapper - Mapping frequency to six different outputs after effector, the pitch number is mapped to pitch, however the freq. number is mapped to all areas other than freq., here that element effects the outputs nominally named duration, inter onset, Velocity/amplitude, and so on.

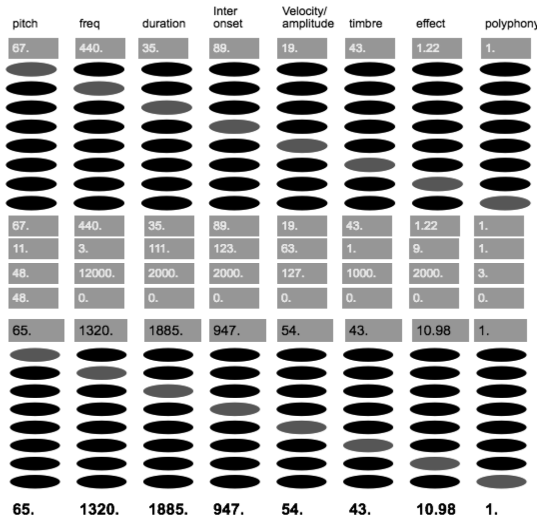

Figure 9: MatrixMapper - Direct map from input to output after effector

The MatrixMapper allows the arithmetic operator discussed above; for example the pitch 67 is multiplied by 11, put through a modulus of 48, the 48 is added to that result, creating a final result of 65. The same processes are put to each of the eight inputs and the varying result are seen.

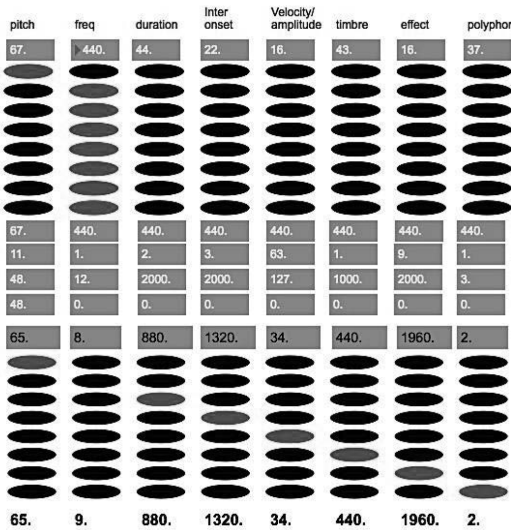

Figure 10: MatrixMapper - Mapping frequency to six different outputs after effector

Polyphony

Five note polyphony based on any sequence of notes is immediately available through the module seen in Figure 11. This process considers the pitch interval between selected notes and applies that interval to up to four additional notes to the original take from the output of the generator, which may be numeric or note based. This process takes the interval between selected notes or numbers and adds that to the next selected note/number. The resulting output can be played at time intervals related to the interonset times of the selected notes, creating arpeggios of varying time.

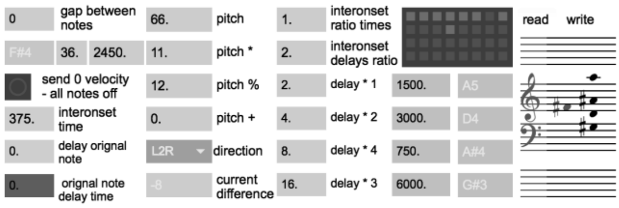

Figure 11: Polyphony – creating a polyphonic playback

Sound playback

Typically sounds can be played back via MIDI systems when well-tempered pitch based systems are the goal. If non-tempered pitches are required this can be accessed through process such as pitch bend when working in MIDI and, frequency mapping when working in audio.

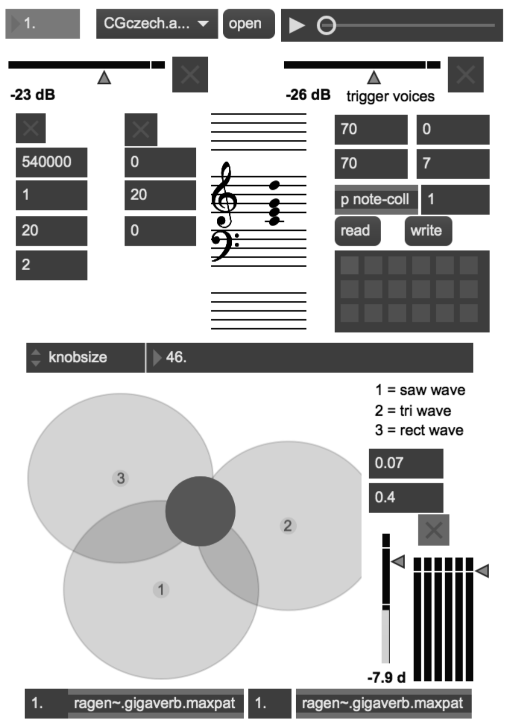

Figure 12: ConSynth – variable saw, triangle and rectangle wave synthesiser

Figure 12: ConSynth – variable saw, triangle and rectangle wave synthesiser, shows a simple-wave synthesiser that convolves any audio input with a mix of those three waves, which can be altered in real time by physical gestures by the composer, or by mapping attributes to data from a NumInput module of the amplitude of either wave form or the mix of all three. Figure 12, shows an iteration of ConSynth in which a sound file unique to that iteration provides the sound source, pitches from a preset sequence of polyphonies creates a harmony, a specific sound file is played, and a mix of saw triangle and rectangle waves set to the frequencies of the four pitches shown are used. These pitches could be selected from the output of a Polyphony module and/or a NoteInput module.

Each of these processes can be reiterated and repeated to create a final work, and the output can be adjusted to form new interpretations via the interface shown in Figure 13 and Figure 14

Interactivity and Provocation

Inside/outside/regular/irregular

It could be said that all improvisation is interactive, be it with the instrument, the listener, other improvisers or with the improvisers intentions. Robert Rowe (1992; 1993; 1999; 2000; 2004) has written extensively regarding systems and created his "own program Cypher [that] analyzes MIDI data on two levels: first, it classifies incoming musical events across a number of features such as loudness, speed, and register; and second, the behavior of these features within the current phrase is characterized as being regular or irregular. The user (the composer) constructs rules using the output of this analysis"(1999). These concepts of regular and irregular are important to ICIA, as they are terms that are contextualized within each composition and performance. They can be synonymous to the ideas of 'inside' and 'outside', and there are degrees of this excursion to and from the expectations of the composer, improviser and listener.

Karlhienz Essl's Real Time Composer (RTC) "offers the possibility to experiment with a number of compositional techniques, such as serial procedures, permutations and controlled randomness. Most of these objects are geared towards straightforward processing of data." (2015). Serial techniques offer a great range of possibilities in all musical styles; an example of this is using the arithmetic approaches used in ICIA and shown in the variety of ways in which the outputs can be mediated, mapped, and translated.

The interactivity built into ICIA is based on the more traditional approaches, such as user input via a score based approach or instrumental input such as performing on a MIDI instrument, user adjustment via the variety of interfaces presented above.

The intricacy of these interactions, where adjusting one parameter can have extensive and potentially unanticipated outcomes provides a provocative workspace for the composer. Simple ideas can be rendered in a wide variety of ways that may not have been immediately apparent to the composer in the initial acts of composing. These may be palatable or not when first heard but may provoke new approaches to composition, listening and aesthetics.

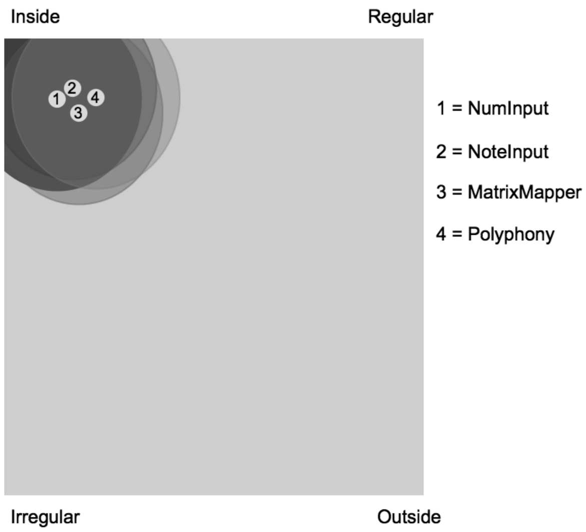

Figure 13: Clumped possibilities for variation

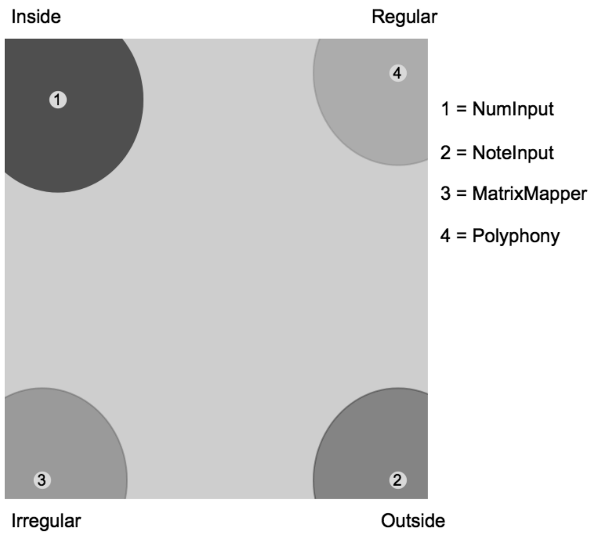

Figure 14: Separated possibilities for variation

The final step in ICIA is the control of all attributes via a cohesive interface that introduces varying order Markov chains to effect the data. Figure 13 shows a situation where each of the four areas used in the composition process will represent the input from the composer without much variation. Figure 14 shows slightly increased variation from the NumInput module, 'regular inside' variation regarding polyphony, 'irregular inside' variation regarding MatrixMapper behaviour, and 'outside regular' variation from the NoteInput module.

Conclusions

"Computer programs are nothing more than human decisions in coded form. … its algorithms based on intuitive subjective preferences."(Roads 2015: 348) and "Music, far more then [visual] art, is the realm where he objective and subjective can be tied together with scientific and logical rigour" (Nelson 2015: 64). Both statements are at the core of ICIA; the program itself is based on the author's intuitive and subjective preferences, and the processes can be seen to have a logical rigour in their systems and subsequent output.

ICIA offers the opportunity to the composer to put in simple ideas and create variations that may not have been intuitively available or within their aesthetic paradigms and anticipations, creating outputs that while directly related to their original choices.

ICIA uses these ideas to create a situation where it is possible to "imagine something you don't know". This is an impossible task as imagination is limited by knowledge. However by beginning form a position based in the imagination of the composer, and through creating a variety of simple options, the outcome may be something that is inherently built on the composers imagination but expressive of something beyond their imagination.

Music created using ICIA can be accessed at https://rogeralsop .bandcamp.com/album/icia/

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[Abstract in Korean | 국문 요약]

작곡가/즉흥연주자의 행동을 대화 형태의 매력적인 작곡시스템에 통합하기

로저 알솝

이 논문은 컴퓨터 기반 자동화된 작곡/분석과 그와 관련된 미학적 이해에 대해 논한다. 맥스Max로 제작된 프로그램 "이샤(통합 작곡 즉흥 액션)ICIA(Integrating Composer Improviser Actions)"는 소리 분석 시스템, 사운드 이펙팅Sound effecting과 조절 시스템, 알고리듬 작곡 프로세스와 행위 인지 시스템으로 구성된다.

"이샤ICIA"는 작곡가/즉흥연주자가 선택한 소리들을 추계적[추측과 통계에 의한]stochastic 과정이 입력된 유저user와 기계를 통해 사용가능한 부분과 그룹들로 나눈다. 이 그룹들은 화면에 띄워져 작곡가/즉흥연주자가 다양한 작곡 과정을 통해 실시간으로 조절될 수 있도록 한다. 작곡가/즉흥연주자가 사용할 수 있는 소리는 환경의 소리, 이미 저장된 소리, "이샤"에 의해 만들어진 소리, 그리고 작곡가/즉흥연주자가 선택하는 모든 종류의 입력 소리를 포함한다.

이는 작곡가/즉흥연주자가 변화무쌍한 조절을 가능케하는 알고리듬 세트를 통해 그들의 신체적, 음향적인sonic, 의도적이거나 비의도적인 행동과 결단을 해석해 나타냄으로써 활기넘치는 환경을 만든다. 이 알고리듬은 작곡가나 기계가 설정하는 '예측가능성 수준'의 범위를 짓는 역할을 한다. '더more'와 '덜less', '안'과 '밖', '규칙'과 '불규칙'의 정도 역시 작곡/즉흥적 과정에 입각하여 사용된다. '비음악적non-musicla'이라는 명령어는 작곡가/즉흥연주자가 전통적인 작곡 시스템과 다른 접근방법을 모색케한다. 이렇게 사운드아트나 음악예술작품을 만들 때 "이샤"와의 상호작용을 통해 여러 방면의 대처를 모색하면서, 작곡가/즉흥연주자가 창작과정에서의 의식적 혹은 무의식적인, 의도적 혹은 비의도적인 미학적 의견을 표출하게 된다.

Interpreting Data:
Re-contextualizing Data to Develop Approaches to Musical Composition

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This paper discusses ways the authors use data from the science in the creation of musical works, *Silver With Carbon Bond*, by Giles and *Flow 2000* by Alsop described below.

Silver With Carbon Bond, created while in-residence at the Bio 21 Institute, is a musical representation of mass spectrometry data. It takes experiment results, which describes the mass to charge frequency, and the true and relative intensities of each frequency of atomic mass of silver bonded with carbon. It relates the mass/charge frequency to specify sound frequency, and the relative intensity to specify amplitude, and a Max patch is used to synthesise output. SPEAR was used to do a spectral analysis and onset detection of the synthesised sounds, and to export that data as SDIF. The SDIF file is then used in a patch developed using IRCAM's OpenMusic that reads the spectral analysis file and outputs notation. This raw notation then formed the basis for the piece Silver With Carbon Bond.

Flow 2000 represents the theoretical behaviour of imaginary fluids as represented through varying the elements of velocity, density, tube length and width, and dynamic and kinematic viscosity. These elements are sonically represented as: frequency, timbre variation based on altering phase of sine waves, positioning within a binaural space, amplitude, and duration. A varying imagined Reynolds number based on randomly generated seed values for the six elements is used to generate the numeric values of these elements.

The paper details the processes used to create the two works, the influence of the composers' aesthetic preferences in the creation of each piece, and potential ways in which the processes may be used in the development of other works that exemplify the music/science nexus.

The intersection between science and music can be an extraordinarily fruitful place to work from, and allows so many approaches to music-making that range from the trivial (such as the fact that a great quantity of music made today utilises computers in some way) to the much more interesting (such as observations made by people like Thomas Watson that sound carries via the ionosphere and can be received by radio (Kahn 2013))! Giles' interest in science is reasonably broad, and stems from an ideal to – as much as possible – objectively understand everything in our environment. Of course, this is not necessarily possible at any given point in our history, but it is an ideal to strive for. More specifically, organised sound[1] as a function that necessarily comes from physics and biological evolution. By necessity, looking for a way to creatively engage with this wider understanding of the world, and thus attempt to understand the world more thoroughly, is important to Giles. This was highlighted when he attended a lecture by composer Richard Barrett, and he discussed the notion that perhaps art (including music) might reflect an intuitive understanding of some natural phenomenon that we are otherwise unaware of: in particular, by observing the role of memory, imagination, and expectation when listening to music and relating that to a possible multi-dimensional (or multi-directional) time, that flows forwards and backwards (Barrett 2014).

Alsop considers the recontextualizing of data from a wide variety of disciplines, modes of thought and systems; looking at the process for gathering, interpreting and using data across many fields in the creation of art works. He initially looked at processes of interpreting speech systems into systems for music/sound art creation (1999; Pedersen / Alsop 2013; Burke / Alsop 2014; Alsop 2014; Alsop 2007; Cox / Alsop 2014; Alsop 2013; 2010; 2011; 2003) creating a number of audio works, and then developed these conceptual and technological processes to generate interactive audio visual works. In these works Alsop developed the algorithms and formulae for interpreting data idiosyncratically and without direct reference to specific or already used approaches.

Discussed in this paper are the computer-based approaches to data re-contextualization employed by Alsop in his work: *Flow 2000* (2015), and by Giles in *Silver With Carbon Bond* (2014-2015). The paper will briefly describe the contexts of these two pieces, explore the techniques used in developing them, and contrast the two approaches, with the expectation that such explorations yield innovation in the arts, and a novel way of exploring data in the field of origin.

Giles in residence: context for the work

Giles is one of six artists-in-residence (and the only composer) for a period spanning approximately 15 months, with an anticipated end date of December 2015. The residency has been facilitated by the Victorian College of the Arts and the School of Chemistry, both faculties within the University of Melbourne, in Victoria, Australia. These works take the form of instrumental and electronic work, some for installation, some for live performance. In particular, Giles has been working with a chemist who specialises in mass-spectrometry[2], as there are some obvious parallels between spectral analysis in sound and mass spectra analysis in chemistry.

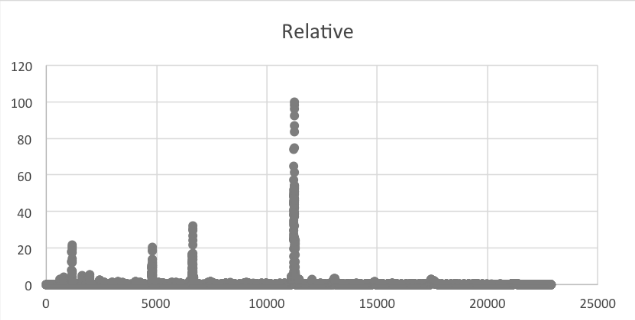

Figure 1. m/z spectra data of silver bonded with carbon. Provided by Zavras (2014).

The work under discussion in this paper: *Silver With Carbon Bond* is for solo baroque violin with either a classical or modern bow; it is a sonically demanding work requiring the performer to execute somewhat complex rhythmic gestures with fine motor control in the bowing arm. It is one of a number of works being developed during the residency. An instrumental piece was decided on, not only to be an interesting challenge, but also an interesting juxtaposition: translating data from mass spectrometry into notation. Not simply being descriptive, but finding novel ways of treating the data as a material basis for the development of a piece that, when looked at in the context of Giles' total output from the residency, can be seen as inter-connected by their source material.

Using the formula for deciding a Reynolds number to create a sound work develops Alsop's conceptual approach discussed above. Here the formula is simply taken as way to develop outcomes to be reified sonically, in this case via live flute performance augmented by technology.

Aesthetic Considerations

The work under discussion as with others (Giles 2015) created by Giles over roughly the last four or so years has forced him to consider an aesthetic stance, and thus context, suitable to partially generative music. This aesthetic stance is built on an intersection of ideas, ideals, values, and a degree of poetics, the primary points of which are detailed below.

The first is an ongoing interest in 'complexity', which is seen as not only musical complexity (complex compared to what? – refer to (Barrett 1992) and (Toop 1993) for in-depth considerations on complexity) but an ever-expanding understanding of complex systems: biological, electronic, social, and so on. This interest in complexity has lead to a great fascination with the 'micro' in sonic works; tiny fragments of sound or material that exist at the edge of human perception (Roads 2001) – a facet of sound that has been at the core of works developed during Giles' Bio 21 Residency.

The second is the recognition of an attitude of 'experimentation' (Cassidy 2012) in musical composition: a practice driven largely by the question 'what happens if…?' and the procedures and works that result from that question (mondayeveningconcerts.org 2011). There are other forms of experimentation also, from systems of constraint to systems of indeterminacy, and so on. But for the most part, experimentation is procedural and generative, and once the music is committed to, it becomes determinate.

These aesthetic considerations may simply be described as the prioritisation of structure and logic, and of unity between material and form, which the above-mentioned ideas and values encapsulate.

Alsop's approach has many similarities to Giles', particularly an interest in the complex systems mentioned above and their behavior. While concepts of artistic frameworks, such as 'the new complexity', 'minimalism', 'abstract' and so on are considered when interpreting his works they are not when creating them. Instead there is a strong focus on improvisation, time, cross-modality and inclusiveness in the creation of the works. These terms may take a variety of interpretations while creating the works; for example, 'improvisation' may refer to the development of an improvised performance, machine improvisation, or improvised interpretation – where the work is listened to and considered through a variety of concepts and contexts, or 'time' may be considered as the temporal relationships between events inherent to the work being made, not considering the work complete until it has been listened to and considered over an extended time period – a year for example. He sees each creation, reification and experience of an artwork as unique to the ephemeral moment of creation, reification or experience, and considers collaboration and improvisation as essential to his approach.

Alsop views understanding per se as a prime human need and that this is created through exploring the structure and logic, materiality and form of things, and when these aspect are interrogated though new, differentiated, or diverse paradigms an aesthetic experience is created that may disrupt or expand current aesthetic norms in both the composer and the listener.

From Data to Notation: Silver With Carbon Bond

In developing the piece titled *Silver With Carbon Bond* (Giles 2014-2015) the composer needed to make tools that could transfer, sonify, and then notate the data that received from the Bio 21 Institute. The goal was to move beyond simple sonification – which could be perceived as somewhat trivial – into artistic territory that thoroughly interrogates the provided data, illuminating a new view of data. The primary generative software used by Giles is MaxMSP (Max) and OpenMusic (OM), along with Sibelius 6 for typesetting.

From data to notation: generation and interpretation

The raw mass spectrometry data is provided as a large Microsoft Excel spreadsheet, consisting of up to ~22,000 individual rows, depending on the sample, and three columns: *m/*z (mass to charge), *absolute intensity* and *relative intensity*[3]. This data plotted as a graph can be seen in Figure 1. The relative intensity data is the most interesting and usable, partly because the numbers are much smaller and thus constrainable, and partly because they are what the scientists prefer to use in their tests.

The first stage was developing software to sonify the data, taking the *m/z* and its relative intensity as frequency and amplitude data in the sound domain. Using a patch designed in Max MSP which performs the job of separating the columns and attaching the *m/z* and relative intensity as a pair, to be sent to oscillators for synthesis. The result of one such sonification can be seen in Figure 2[4]. This process is fundamentally a gradual additive sequence of one oscillator being assigned a frequency and amplitude, then the next, and so on.

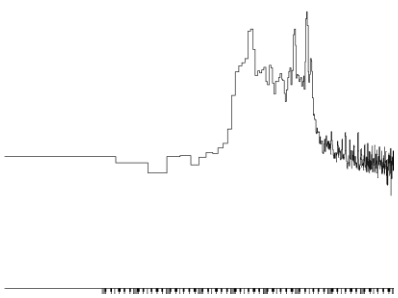

Figure 2. Cumulative frequency data generated using MaxMSP.

This generated audio is the most rudimentary of materials with which to work, acting as the unifying basis from which all other materials are derived[5]. SPEAR[6] provided the functionality of being able to perform spectral analysis on the source audio and to output an SDIF[7] file that can then be used within OM. The SDIF file is the source of generative pitch and rhythm, with pitch being approximated to any division of the octave that is desired: in this case, 1/4 tones or the division of one octave into 24 equal-tempered (24TET) parts. Once this data is within the OM environment it exists in the form of lists, which have the flexibility of being subjected to variation of any degree of complexity through procedures that reflect those employed by serial composers: inversion, retrograde, and so on, alongside other manipulations such as randomized rhythm or rest insertion, polyrhythmic nesting of tuplets, and much more. Given the quantity of data (and subsequent material that would be generated), no such permutations or manipulations were employed other than quantizing the onsets as rhythms to a 13-tuplet at any rhythmic division of a quarter note (8^{th}, 16^{th}, 32^{nd}, 64^{th}, and so on). Figure 3 (below) shows the patch and its generated pitch and rhythmic material.

Once quantized, the rhythm part became stretched well beyond the time of the original audio, which led to the decision to make the piece – for solo baroque violin with classical or 17^{th} century bow – in four short sections, each corresponding to a part of the parent audio. In this way, the piece will coalesce in an additive way, similar to the sonification process, resulting in the final section that contains as much of source sonic information, spread over time, as is possible on a fundamentally monophonic instrument.

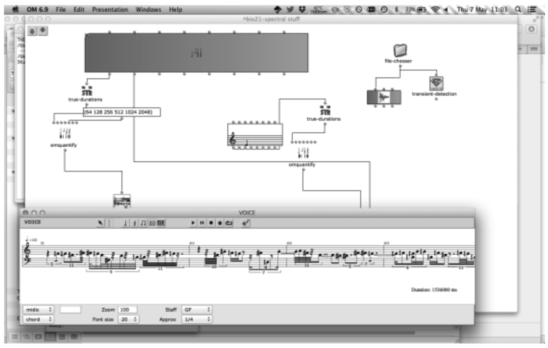

Figure 3. Combined pitch and rhythm generation using OpenMusic.

Having generated the raw material, the actual composing could begin. This process involved taking the raw notation generated by OM and getting it into the typesetting software Sibelius[8], and then working with that raw material to turn it into a piece of music. Figure 4 (below) shows the first page of the completed, typeset composition with much more detail, rhythmic division and formal organization, various performative parameters such as glissandi, bow pressure, and harmonics, and so on.

This process of working with generative material forces a response to the material that is intuitive, and becomes almost like a performer responding to a score without the restriction of time. It stimulates the imagination and forces, as Cassidy (Cassidy 2013) discusses, the composer into a metaphorical corner from which they must push against the imposed restraints, stimulating imagination and invention that would otherwise not occur. In this sense, this method of composing can be considered experimental, in that the outcome is not known due to the generative procedures, but once those procedures have run, it is up to the composer (and then the performer) to respond.

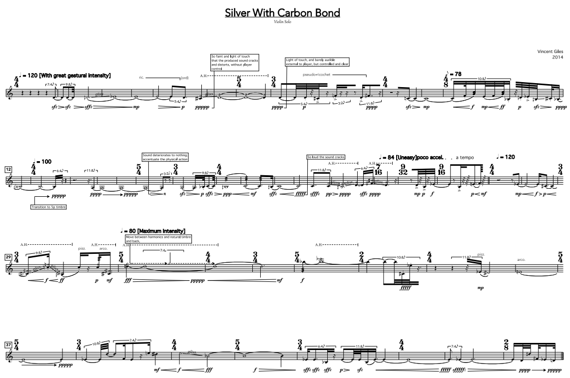

Figure 4. Score for first page of Silver With Carbon Bond. (Giles: 2015)

From Data to Notation: Flow 2000

The data used to create Flow 2000 is imaginary; the process of using the Reynolds number equation to create a sonic output offers a very different approach to Giles, who took existent data from an analysis of an object to for a basis for his composition. The Reynolds number equation has a variety of iterations depending on the properties of the object being tested, as this version of its use is to be performed with a flute the iteration based on pipes is used, this can be seen in the equation $R = \frac{\rho v D}{\mu}$ (Ahern 2007), this variation of the equation $R = \frac{v\rho L}{\eta}$, which provides "a more general approach to turbulence when objects move through a fluid" (Nave 2012) does not form a substantive difference in outcome when used in the creation of sound.

The equation provides ten potential numbers to be generated, one for each symbol and one for the inverse of the outcome of the equation. These numbers provide the score as is shown in Figure 5.

This is a dynamic score, created in the Max environment, that responds to the performer and performance as it is taking place. The performer is able to interpret the score within a few paradigms; for example: air pressure 1 - 10 refers to the pressure the performer should apply to exciting the playing the note indicated below, the angel of embouchure indicates the position the performer should angle the flute in relation to their mouth, key depression indicates the amount to which the performer should depress the keys in order to play the note, and units of time refer to the number of beat the performer should take to play the particular note. The performer can set the unit of time prior to performance in relation to their experience of the dynamics and reverberation time of the room.

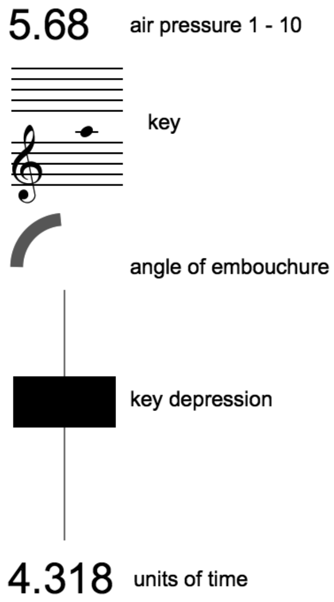

Figure 5: Flow 2000 score example

The music made is generated from using the $R = \frac{\rho v D}{\mu}$ equation based on the frequency of a flute note (in this example a recording of a flute performance was used to generate the data), as shown in **Figure 6**.

	reynolds equation		inverse of reynolds equation
v *	785.963013	v /	620472.920259
p *	784.475891	p /	622538.467483
D /	787.087402	D *	617451.091338
n =	780.655334	n =	611534.435264
R	621649.143367	! R	0.987131
	reynolds number		

Figure 6. Reynolds equation based on changing frequencies

These numbers are mediated to fit the ranges appropriate to the score element displayed, for example the relating to *v* s scaled t fit between 1 and 10, the number relating to *p* scaled to fit between integers 60 and 96 so that is can be represented as a MIDI note, the number relating to *D* scaled to fit between integers 0 and 180 to represent an angle, the number relating to *n* scaled to fit between integers 0 and 100 to relate to percentages of depression of the keys, an the number relating to *R* multiplied by 0.00001 to indicate a duration. It is understood that any performer and performance venue may require alteration to the scaling process and this is built into the Max patch.

The inverse of the resulting number, created by reversing the operations used to get the Reynolds number, is used to effect the reverb time, bandwidth, damping, early reflections, and reverb tail time parameters of the gen~ gigaverb patch. This creates a virtual sonic space that responds to the performance in situ. The entire control surface is seen in Figure 7, this presents all relevant data but only the score, seen in Figure 5 is visible to the performer.

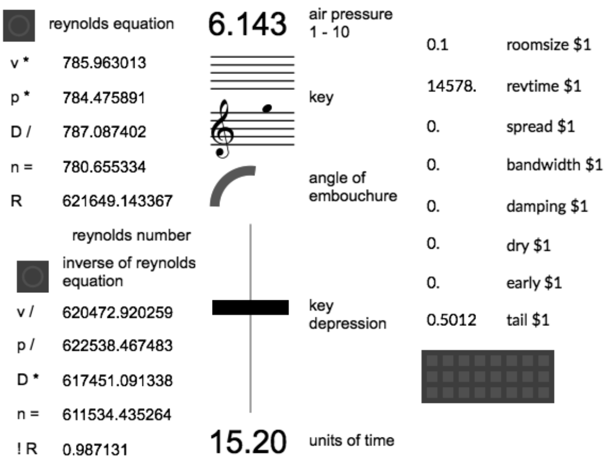

Figure 7. Flow 2000 control surface

Flow 2000 integrates Alsop's interests in collaboration, improvisation, the ephemeral nature of sound art and performance, and the importance of location in the creation of an artwork.

Comparison of Approaches and Concluding Remarks

Data is often seen as exclusively the domain of the sciences, understandable only in the context from which it comes or which it describes. However as both Giles and Alsop have shown, data interpretation can be very fruitful ground for artistic expression in musical activity and the sonic arts, providing the artist (composer) moves beyond sonification and into interpretation.

The aesthetic considerations and goals of Alsop and Giles are different, as are their approaches to data interpretation. Giles prefers an 'experimentation-as-material-generation' approach, which once generated becomes extremely deterministic at a performance level, while Alsop takes an interactive approach where the performer becomes part of a pre-designed system of data interpretation with indeterminate outcomes. Both approaches reflect an experimental approach to music making that is based on intransigent approaches, generated data in Giles' case and a mathematical formula in Alsop's.

One novel connection that arises between the two approaches is Alsop's notion of a virtualised sonic space. In Alsop's approach the virtualised sonic space exists within a computer system, and is both a score-generator and performer-interpreter, that exists only in the virtual world. The Reynolds number, which is dimensionless and at best creates an indication of an outcome, provides a metaphor for the constant fluid motion of a performance.

The score in Giles' approach appears to function in a similar role, being the artefact around which the virtualised sonic space (the data interpretation systems built by Giles) and the performer interact – a sonic mediator of sorts.

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[1] *Organised Sound*: acts as an umbrella term for sound art, sonic art, music, and so on. I will interchangeably use the terms *sound art* to mean work in which the visual element and the sonic element in an installation context are equally important; *sound art* will also encompass radiophonic work. *Music* will be used to describe works which come from the Western art music (classical) tradition, including variants such as: computer music, acousmatic music, electroacoustic music, and so on.

[2] Electrospray ionization mass spectrometry (ESI-MS) is an analytical method allowing transfer of molecules in the liquid-phase to the gas-phase where they can be analysed according to their mass-to-charge ratio (*m/z*). The ionization process (ESI) can transform neutral molecules into charged species (ions) classified as positively charged cations or negatively charged anions. The mass analyser allows the mass of the molecule to be revealed by information in the *m/z*. The analysis produces a mass spectrum that displays the relative intensity (Y-axis) vs. the *m/z* (X-axis).

[3] Relative intensity is the electron charge per spectra relative to the sample currently being tested.

[4] Audio of this wave file is available here: https://soundcloud .com/vince-giles/firstsynth7000/

[5] Indeed, this method of 'material restriction' has been part of my composition practice for many years. It provides a way of unifying a piece across any amount of time, and is for me a great way of inducing creativity. It means that the starting material can be explored in great detail over the course of one or many pieces of music. In this case, the parent material – the data – becomes the unification across multiple works, while at the localised level of *Silver With Carbon Bond,* only the generated waveform acts as the unifying material.

[6] *SPEAR*: Sinusoidal Partial Editing Analysis and Resynthesis. http:// www.klingbeil.com/spear/

[7] *SDIF*: Sound Description Interchange Format. http://sdif .sourceforge.net/

[8] Giles has since switched to Finale, making the process of porting from OM to typesetting much quicker. At this stage however, the OM output needed to be transferred to Finale Notepad to be re-exported as an older format of MusicXML that Sibelius could recognize.

[Abstract in Korean | 국문 요약]
데이터 해석: 작곡에 유용하도록 데이터 재맥락화하기

로저 알솝 / 빈센트 길레스

이 글은 두 저자의 음악 작품, 길레스Giles의 《탄소결합 은Silver With Carbon Bond》과 알솝Alsop의 《플로우 2000 Flow 2000》을 제작할 때 과학적인 데이터를 활용한 방법에 대해 논한다.

《탄소결합 은》은 저자가 바이오21Bio21 연구소의 상주작곡가일 때 만든 질량 분광분석spectrometry 데이터를 음악적으로 구현한 작품이다. 이 작품에서 질량 대 전하의 빈도frequency와 탄소결합 은의 원자질량이 갖는 절대적, 상대적 강도의 빈도수를 나타낸 실험 결과들을 다룬다. 이 결과들은 질량 및 전하의 빈도수를 소리의 주파수로, 상대적 강도를 음량으로, 맥스패치Max Patch는 출력을 합성화하는데 관계지어 사용된다. 스피어SPEAR는 분광분석을 하거나, 합성사운드를 개시 탐지onset detection 하고, 그것을 에스디아이에프(소리묘사교환형식)SDIF(Sound Description Interchange Format) 데이터로 추출하는 데 사용되었다. 이 추출된 에스디아이에프 파일은 이르캄IRCAM의 오픈뮤직OpenMusic 기반의 패치형태로 분광분석 파일을 해석하여 그 결과를 기보해낸다. 이렇게 그려진 악보raw notation가 《탄소결합 은》의 기초가 되었다.

《플로우 2000》에서는 상상의 유연한 물체가 속도와 밀도, 튜브모양의 길이와 폭, 점성의 역동성과 운동성의 [여섯] 요소들이 변화하면서 표현되는 이론적인 행위를 그려낸 작품이다. 이 요소들은 소리의 여러 음높이, 사인파의 위상변화로 인한 다양한 음색, 공간적 위치변화에 의한 다양한 스테레오binaural 효과, 셈여림과 길이 부분으로 전환되었다. [앞서 언급한] 여섯 요소의 임의로 생성된 값에 기반하여 여러가지 상상의 [액체의 흐름을 나타내는] 레이놀즈수Reynold numbers로 이 부분들의 수치를 결정하였다.

이 글은 두 작품을 창작하는 과정과 각 작품에서 나타나는 작곡가의 효과적인 미학적 선택, 앞으로 음악/과학 융합작품에서 사용가능한 방법들을 세세히 기술한다.

Rccm Canons:
a real-time spatialization algorithm

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Regular Complementary Canons of Maximal Category (RCCM) or *Vuza canons* were introduced to the musical world with Dan Tudor Vuza's seminal papers in PNM in the 1990s. Since their original exposition, this important notion has known many developments and generalizations.
Musicians have always been intrigued by canon construction, q.v. the complex polyphony of the Flemish composer Josquin Desprez or the contrapuntal techniques of Johann Sebastian Bach, whose properties have been translated into formal algebraic terms. Olivier Messiaen is maybe the first theorist and composer who introduced and studied the concept of rhythmic canon. From a mathematical point of view, the construction of rhythmic canons is formalized in terms of factorizations of finite cyclic groups as the sum of two subsets, a concept that probably Messiaen, like many other composers of the twentieth century, did not know. Nowadays the process of building mosaic canons can be implemented using programming languages: composers can access extremely complex macro-structures that can be taken as the basis for formal architectures of original music compositions. In this paper, we would like to show our implementation in Wolfram Mathematica, with an algorithm, that moves from the Franck Jedrzejewski one, which allows the composer to create and directly manage RCCM Canons. In addition, we use our algorithm to construct special patterns for real-time spatialization.

A canon is a recognizable pattern that is repeated with different offsets, typically with different instruments or simply different voices. This pattern, called *motif* or *inner voice*, can be modified (i. e. augmented or retrograded). In this paper, we will take into account the construction of a particular type of canon, the rhythmic one.

When we study *rhythmic* canons, we only need to consider their rhythmic properties (i.e. the occurrences of musical events, regardless of pitch, timbre or dynamics). A rhythmic profile acts as a *tile* that the composer tries to translate temporarily in order to construct macro-structures having the properties to fill the entire space of the pulse, without silence between a rhythmic pulsation and another, and without superpower position among the various voices, which are therefore mutually complementary. A rhythmic canon of this type is also called *mosaic* since it creates a regular tiling of rhythmic space.

The geometric properties of these entities, after more than twenty years from their exposition, are still a topic of great relevance for mathematicians and musicians.

Musicians have been always intrigued by canon construction, q.v. the complex polyphony of the Flemish composer Josquin Desprez or the contrapuntal techniques of Johann Sebastian Bach, whose properties have been translated into formal algebraic terms. Olivier Messiaen is maybe the first theorist and composer who introduced and studied the concept of rhythmic canon. In some of his compositions, for example *Visions de l'Amen* (1943) for two pianos or *Harawi* (1945) for piano and voice, the use of rhythmic canons anticipates some formal features of mosaic music, although he did not give a formalization of this rigorous compositional process.

From a mathematical point of view, the construction of rhythmic canons is formalized in terms of factorizations of finite cyclic groups as the sum of two subsets, as we will later see, a concept of which Messiaen, like many other composers of the Twentieth century, was probably unaware of. An exception is surely the composer and engineer Iannis Xenakis who used mathematical tools for the development and the formal organization of the compositional material.

Today the situation is quite different thanks to the development of music informatics. In fact, the process of building mosaic canons can be implemented: the composer can access extremely complex macros-structures that can be taken as a basis for formal architectures of original musical compositions.

Rhythmic canons

As we said, a canon is a polyphonic musical form born in the XIV century. It is a contrapuntal compositional technique or texture that employs a melody with one or more imitations of the melody played after a given duration. If we choose a motif A, transformations τ_i, $i \in I$, that may be only offsettings (viz. translations in time), a canon is the reunion $\mathcal{C}$ of the transforms $\bigcup_{i \in I} \tau_i(A)$. If a motif is identified with its characteristic function 1_A : $D \to \{0,1\}$, with D which is a subset of integers $\mathbb{Z}$, then the superimposition of all its copies appears as a sum: $\mathcal{C} = \sum_i 1_{\tau_i(A)}$.

Most musical canons in the classical sense have nil to many notes on each beat. However, it is possible to specialize more interesting kinds of canons; we distinguish them in *coverings* and *packings*. The first type is a canon where every available beat features at least one note, maybe more ($\forall t \in D, \mathcal{C}(t) \geq 1$). The second one, instead, is a canon where there is never more than one note on every possible beat ($\forall t \in D, \mathcal{C}(t) \leq 1$). Considering the entire problem from a mathematical point of view, it is interesting to define it asking one and only one note per beat; this is called *tiling* (i.e. a mosaic with copies of one motif) maybe allowing some deformations. We can write our problem as:

$$\boldsymbol{tilings = coverings \cap packings : \forall t \in D, \mathcal{C}(t) = 1}$$

How to define a rhythmic canon

Let $\boldsymbol{G}$ be a finite *cyclic group*[1], and $\boldsymbol{S}, \boldsymbol{R}$ two non-empty subsets of $\boldsymbol{G}$. If each element $\boldsymbol{g} \in \boldsymbol{G}$ can be expressed uniquely in the form $\boldsymbol{g = s + r}$ with $\boldsymbol{s} \in \boldsymbol{S}$ and $\boldsymbol{r} \in \boldsymbol{R}$ then the equation $\boldsymbol{G = S \oplus R}$ is called a *factorization* of $\boldsymbol{G}$. A rhythmic canon of $\boldsymbol{G}$ is a factorization of $\boldsymbol{G}$ in two subsets. The subset $\boldsymbol{S}$ is called the *inner voice* or *motif*, while $\boldsymbol{R}$ is called the *outer voice*. If $\boldsymbol{G}$ is the cyclic group $\mathbb{Z}_{\boldsymbol{N}} = \{\boldsymbol{0, 1, 2, \dots, N-1}\}$, the set $\boldsymbol{S}$ can be seen as a tile of the line $\mathbb{Z}_{\boldsymbol{N}}$ and we said that $\boldsymbol{S}$ tiles $\mathbb{Z}_{\boldsymbol{N}}$ with $\boldsymbol{R}$ or that $\boldsymbol{R}$ tiles with $\boldsymbol{S}$, since if $(\boldsymbol{S}, \boldsymbol{R})$ is a rhythmic canon, also $(\boldsymbol{R}, \boldsymbol{S})$ is a rhythmic canon. We can say that tiling the line is the same as finding a decomposition of the generic cyclic group $\mathbb{Z}_{\boldsymbol{N}}$.

A non-empty set $\boldsymbol{S}$ is *periodic* if there is an element $\boldsymbol{g}$ of $\boldsymbol{G}$ such that $\boldsymbol{S + g = S}$ and $\boldsymbol{g \neq 0}$ (the identity element of $\boldsymbol{G}$). In 1948, the Hungarian mathematician György HAJÒS hypothesized that in a factorization of $\mathbb{Z}_{\boldsymbol{N}}$ in two factors one of the factors had to be periodic. This conjecture is false, and the counterexamples are exactly Vuza canons.

Vuza canons

A factorization of a cyclic group is said to be trivial if at least one of its factor is periodic. In the 1950s, Hajòs proved that it is not necessary for one of the two subsets to be periodic. Between 1941 and 1957, mathematicians such as Nicolaas Govert DE BRUIJN, László RÉDEI and the same Hajòs showed several examples of groups for which this is true.

In the 1990s, Dan Tudor VUZA published four articles devoted to the formalization of a particular class of rhythmic canons: RCCM canons.

To define a Vuza canon, let's start with some definitions.

Definition 1. Let G be an Abelian group. A *k-factorization* of G is a factorization in the direct sum of k of its elements.

A *k-factorization* of $G = A_1 \oplus A_2 \oplus \dots \oplus A_k$ is *periodic* if exists an index $i \in \{1, 2, \dots k\}$ so that A_i is periodic.

A *k-factorization* which is not periodic is said *aperiodic*.

Definition 2. Let G be an Abelian group.

G is *k-Hajòs* if every of its *k-factorization* is periodic.

G is *not k-Hajòs* if exists at least one of its *k-factorization* which is aperiodic.

For $k = 2$ we will simply say *of Hajòs* or *non-Hajòs*.

Definition 3. Let G be a finite cyclic group and S, R two non-empty subsets of G. A *Vuza canon* (S, R) is a rhythmic canon

$$\boldsymbol{G = S \oplus R} \qquad (2)$$

where neither S nor R is periodic. In other words, a Vuza canon is a *2-factorization* of a cyclic group. The order N of the group is the canon *period*.

Now, we are going to enounce some theorems that are necessary in order to correctly implement the algorithm for the creation of RCCM canons. The proof of each of them is given in many group theory textbooks.

Theorem 1. The following statements are equivalent:

(i) Vuza canons only exist for orders N which are not in the form

$$\boldsymbol{p^{\alpha}\ (\alpha \geq 1), p^{\alpha}q\ (\alpha \geq 1), p^2 q^2, p^{\alpha}qr\ (\alpha = 1, 2), pqrs} \qquad (3)$$

where p, q, r, s are different primes.

(ii) Vuza canons only exist for orders N of the form

$$\boldsymbol{N = n_1 n_2 n_3 p_1 p_2} \qquad (4)$$

where $\mathrm{p}_1, \mathrm{p}_2$ denote different primes, $\mathrm{n_i p_i} \geq 2$ for $\mathrm{i} = 1, 2$ and $\mathrm{n_1 p_1}$ and $\mathrm{n_2 p_2}$ are coprime.

This first theorem shows some of the essential conditions to find a period for a Vuza canon.

Theorem 2. Let G be a finite cyclic group and S, R two non-empty subsets of G. The following statements are equivalent.

(i) The sum $S + R$ is direct and is equal to G.

(ii) $G = S + R$ and $|G| = |S||R|$.

Theorem 3. Let p_1, p_2 denote different prime numbers, and n_1, n_2, n_3 be positive integers such that the product $n_1 p_1$ is coprime with $n_2 p_2$, then the pair (S, R) defined by

$$\boldsymbol{S = A + B} \qquad (5)$$

$\boldsymbol{R} = (\boldsymbol{U} + \boldsymbol{V'} + \boldsymbol{K_1}) \cup (\boldsymbol{U'} + \boldsymbol{V} + \boldsymbol{K_2})$ (6)

is a Vuza canon of $\mathbb{Z}_N$ with $N = n_1 n_2 n_3 p_1 p_2$ if R is a non-periodic subset of $\mathbb{Z}_N$ and:

$A = n_1 n_3 p_1 \mathbb{I}_{n_2}$
$B = n_2 n_3 p_2 \mathbb{I}_{n_1}$
$U = n_1 n_2 n_3 p_1 \mathbb{I}_{p_2}$
$V = n_1 n_2 n_3 p_2 \mathbb{I}_{p_1}$
$U' = n_2 n_3 p_1 \mathbb{I}_{p_2}$
$V' = n_1 n_3 p_1 \mathbb{I}_{p_1}$
$K_1 = \{0\}$
$K_2 = \{1, 2, \dots n_3 - 1\}$

All these conditions allow us to set all the parameters once we find the five numbers n_1, n_2, n_3, p_1, p_2.

We remember that $\mathbb{I}_j$ is a set defined as $\mathbb{I}_j = \{0, 1, \dots j-1\}$; e.g. if $n_1 = j = 3$, than $\mathbb{I}_{n_1} = \{0, 1, 2\}$.

Theorem 4. Let p_1, p_2 be different prime numbers, and n_1, n_2, n_3 be positive integers such that the product $n_1 p_1$ is coprime with $n_2 p_2$ and $n_1 p_1$ is coprime with $n_2 p_3$. Let H be the subgroup $H = p_2 \mathbb{I}_{n_1 n_2 n_3}$ of $\mathbb{Z}_N$ with $N = n_1 n_2 n_3 p_1 p_2$, and K be a complete set of *cosets*[2] representatives for $\mathbb{Z}_N$ modulo H such that $K = K_1 \cup K_2$; then the pair (S, R) defined by

$\boldsymbol{S} = \boldsymbol{A} + \boldsymbol{B}$ (7)

$\boldsymbol{R} = (\boldsymbol{U} + \boldsymbol{V'} + \boldsymbol{K_1}) \cup (\boldsymbol{U'} + \boldsymbol{V} + \boldsymbol{K_2})$ (8)

is a Vuza canon of $\mathbb{Z}_N$ if R is a non-periodic subset of $\mathbb{Z}_N$ and:

$A = n_1 p_1 p_2 \mathbb{I}_{n_2}$
$B = n_2 n_3 p_2 \mathbb{I}_{n_1}$
$U = n_1 n_2 p_1 p_2 \mathbb{I}_{n_3}$
$V = n_1 n_2 n_3 p_2 \mathbb{I}_{p_1}$
$U' = n_2 p_2 \mathbb{I}_{n_3}$
$V' = n_1 p_2 \mathbb{I}_{p_1}$

This last set is the one we implemented in our algorithm.

The algorithm

Now it is possible to proceed to the algorithmic implementation using *Mathematica*. It is a programming language developed by *Stephen Wolfram*, which has a powerful and intuitive management of lists and a myriad of features already implemented, some of which were useful for this algorithm.

The program takes as input a value called *targetNumber* that establishes the maximum value within which Vuza periods will be generated. After the period list has been created, the user enters in *chosenPeriod* the desired Vuza period and the algorithm generates all the *quintuples* (the five numbers n_1, n_2, n_3, p_1, p_2) related to the chosen Vuza period. Finally, the user chooses the preferred quintuple then *R* and *S* are generated.

Analysis of the implementation of the algorithm

The first step is to use the function *genPrimes* that generates prime numbers from 2 up to *targetNumber* (as clarified above, *targetNumber* is a user-specified value that influences the number of the generated Vuza periods).

```
genPrimes[targetNumber_] :=
  Module[{pList = {}, i},
    (i = 1; While[Prime[i] < targetNumber,
      (AppendTo[pList, Prime[i]]; i++)];
     Flatten[pList])];
```

Then, we know from *Theorem 1* the criteria that a period N (hereinafter «n») has to verify in order to be a Vuza good period:

- $n = p^\alpha$ or $n = p^\alpha q$ with p and q prime numbers and α natural number between 1 and 10
- $n = pqr$ with p, q e r prime numbers
- $n = pqrs$ with p, q, r e s prime numbers
- $n = pqr^2$ or $n = p^2 q^2$

So, we need to create some lists that contain all the periods that do not satisfy these conditions, then we make the complement.

These lists are named *checkList1*, *checkList2*, *checkList3* and *checkList4* and the complement operation is made by the *checkHajos* function.

The *checkHajos* function calls inside itself another function called *checkNumber*: given a number called *num* and a *checkList*, using the *Cases* function (natively implemented by Mathematica), *checkNumber* returns the number only if it is not present in the considered *checkList*. The *checkHajos* function establishes that the provided number is not present in any of the checklists.

Giving as input a range of numbers from 2 to *targetNumber*, the *checkHajos* function will return all the Vuza good periods that are equal or less than *targetNumber*.

In the next part of the algorithm the user chooses a Vuza period, called *chosenPeriod*, and it must be factorized in such a way as to produce quintuples of numbers.

The *FactorInteger* function factorizes in the following way:

{{n1_base,n1_exponent},{n2_base,n2_exponent}, ...}.

Using the *extendedFact* function, it is possible to transform the list *{base, exponent}* into a list that contains the individual factors without using exponents.

Applying the previous function to all the sublists in the form of *{base, exponent}* provided by the function *FactorInteger* it is possible to obtain a list containing all the individual factors with exponent equal to 1 (i.e. in the case of the Vuza period N = 72 the function FactorInteger returns the list {{2, 3},{3, 2}} and the function extendedFact applied to the two sublists returns {2, 2, 2, 3, 3}).

In order to verify which elements of the quintuple will be n_1, n_2, n_3, p_1, p_2, it is needed to check that $n_1 p_1$ is coprime with $n_2 p_2$ using the *CoprimeQ* function.

In addition the following conditions must be met: $p_1 \neq p_2$ and $n_1 + n_2 + n_3 \geq 2$.

Applying the function to all the possible permutations of the quintuple, the list of the possible Vuza quintuples, sorted this way *{n1, p1, n2, p2, n3},* will be returned.

At this point, the user must choose a quintuple, called *chosenQuintuple*, among those generated by the previous function. After that it is possible to calculate S and then R (from equation 7 and 8).

Firstly, we need to generate A and B (denoted in the code with a and b):

```
a = Table[i * n1 * p1 * p2, {i, 0, (n2 - 1)}];
b = Table[i * n2 * n3 * p2, {i, 0, (n1 - 1)}];
```

The list named a has length n_2, while the list named b has length n_1; a is given by the multiplication of an i variable, ranging from 0 to $(n_2 - 1)$ multiplied by $n_1 p_1 p_2$. Something similar happens with the computation of b, but it ranges from 0 to $(n_1 - 1)$ and is multiplied by $n_2 n_3 p_2$.

Then the additions of two values of each element of a with each element of b are calculated and assigned to a list called *points*.

The previously chosen Vuza period is now appended at the end of the list and the list is flattened.

Now we need to calculate the differences between the elements contained in the *points* variable using the *Mathematica* natively implemented function *Differences* in order to obtain S in its *basic form*, i.e.: if $S = \{\alpha, \beta, \gamma, \delta, \varepsilon, \vartheta\}$ with $\alpha, \beta, \gamma, \delta, \varepsilon, \vartheta \in \mathbb{Z}_N$, then its basic form is given by $S_B = \{\beta - \alpha, \gamma - \beta, \delta - \gamma, \varepsilon - \delta, \vartheta - \varepsilon, N - \vartheta\}$. S represents the entry interval of each voice of the canon.

Now it is time to calculate R, using a similar strategy to the one used for S; R represents the rhythmic pattern of the single voice, a pattern that is repeated after a certain entry interval contained in the S list.

To find U, V, U', V', we define $K_1 = 1$ and $K_2 = 0$. These are denoted in the code as $u, v, u1, v1, k1, k2$.

```
u = Table[i * n1 * n2 * p1 * p2, {i, 0, (n3 - 1)}];
v = Table[i * n1 * n2 * n3 * p2, {i, 0, (p1 - 1)}];
u1 = Table[i * n2 * p2, {i, 0, (n3 - 1)}];
v1 = Table[i * n1 * p2, {i, 0, (p1 - 1)}];
```

Then it is necessary to compute all the possible additions of two elements between the elements of $v1$ and u and to add $k1$.

$u1vk2$ is then calculated in a similar way, except that the value of the constant $k2$ is equal to 0.

R is obtained from equation 8, flattening the resulting list. In order to obtain R in the *basic form,* it is needed to apply the *Differences* function.

We would like to underline how the algorithm is able to calculate all the quintuples with maximum efficiency and then all the possible R and S and it outputs R and S in the absolute and basic (or relative) form.

It is also important to underline that S and R are interchangeable and that R can become S and vice versa.

Some results

We generated both graphs and sounds. In this section, we will show some representations of a Vuza canon with period 264 using the quintuple {11, 3, 2, 2, 2}.

In this case R in the basic form is *{1, 3, 19, 22, 43, 4, 41, 22, 21, 1, 3, 84}* and S in the basic form is *{8, 8, 8, 8, 8, 8, 8, 8, 2, 6, 2, 6, 2, 8, 8, 8, 8, 8, 8, 8, 8, 118}.*

In the following pictures, it is possible to see the voices of the canon on the Y-axes and time on the X-axes expressed in seconds.

We are plotting a Vuza canon, so, we obtain, as we should, only one point per X coordinate for each instant.

In Figure 1, from ninety to one-hundred-twenty seconds, it is possible to see eighteen voices playing together.

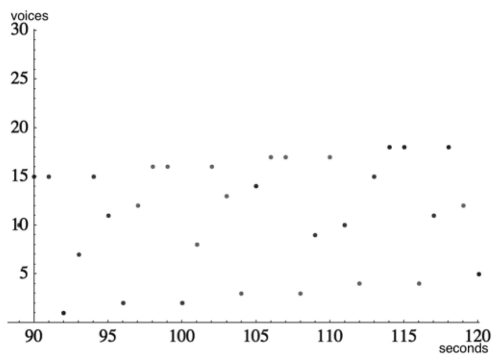

Figure 1. Eighteen voices of the Vuza canon with N=264.

In Figure 2, it is possible to see the end of the Vuza period, 264, for the first voice of the canon.

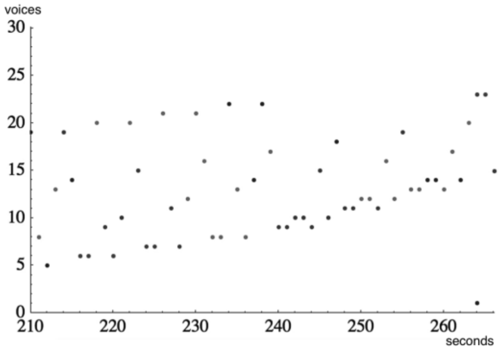

Figure 2. The first voice of the Vuza canon with N=264 at its end.

Real-time Sound Spatialization

New technologies offer many possibilities to composers working with the physical movement of sound, such as, for instance, the trajectories followed by sound events through space, continuous modulation on harmonic and dynamic levels, and various types of proliferations of sound layers. As Berio asserted, the most interesting possibilities are not

> these situations in themselves, but rather the relationships that are established between such physical–acoustic sound mobility and the effective mobility of the musical thought. (Giomi, F. / Meacci, M. / Schwoon, K. 2003)

We agree that computer and sound diffusion technologies enable to live new, unusual acoustic spaces, making even traditionally closed spaces versatile and accessible. This brings to the alteration of the listening perspective and to the creation of a multiplicity of sound levels in continuous transformation.

To reach this aim, we chose Vuza canons to generate patterns for real-time sound spatialization. We have created a system that associates RCCM parameters to the sound locations as sequences of configurations of loudspeakers. The system can be applied for a variable number of sound sources to be moved. It consists of a series of engines that operate on the amplitude of the audio signal, redirecting it toward a variable number of outputs according to the data obtained thanks to the *Mathematica* implementation shown above.

We use MAX/MSP environment to manage the reproduction of sounds and the sound spatialization system. As we hear sound in the form of power, we have chosen to use the equal panning power law, which means that we perceive the dynamic level of audio signals as the square of the input amplitude. To obtain an equal and smooth panning, a linear one, it is necessary to use the inverse of the square operator, which is the square root. Obviously, the square root is not a linear function but the important point is that we perceive the operation imposed on the signal to be linear, as the square is neutralized by the square root, rendering a linearly changing pan when turning the pan pot linearly.

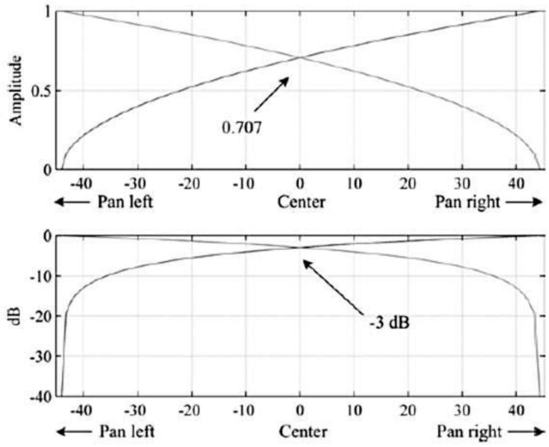

Figure 3. Equal power panning with root of the amplitude (top) and dB of the root amplitude (bottom). Image from "Introduction to digital signal processing" by Tae Hong Park – World Scientific Press 2009.

The equations we implemented are:

$$y[n]_{speaker1} = \sqrt{x[n]}$$

$$y[n]_{speaker2} = \sqrt{1.0 - x[n]}$$

where $y[n]$ is the amplitude of the considered speaker of $x[n]$ input signal.

We applied this algorithm to an octophonic sound diffusion system. R and S obtained by Vuza canon implementation are used to define the active speaker and the transition time from one speaker to another.

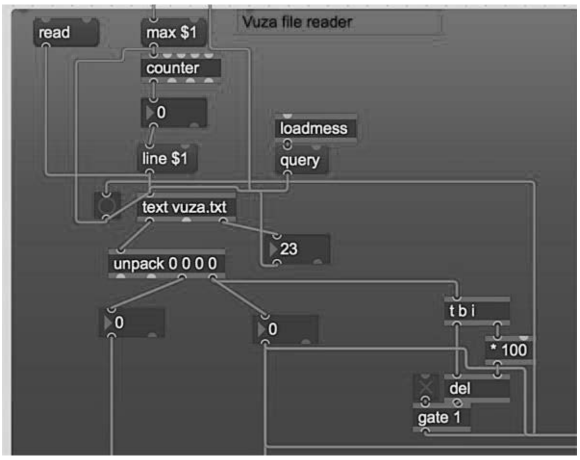

Figure 4. From the MaxMSP environment: the code reads R and S from a text file.

The MaxMSP code reads the selected R and S from a text file as shown in figure 4. The *counter* object keeps counting based on bang messages to read each line of the file. An example of the output file produced by Mathematica is shown in figure 5.

```
1 0 0
2 1 8
3 4 16
4 23 24
5 45 32
6 88 40
7 92 48
8 133 56
9 155 64
10 176 66
11 177 72
12 180 74
13 264 80
14 0 82
15 1 90
16 4 98
17 23 106
18 45 114
19 88 122
20 92 130
21 133 138
22 155 146
23 176 264
```

Figure 5. An extract from the output text file by Mathematica.

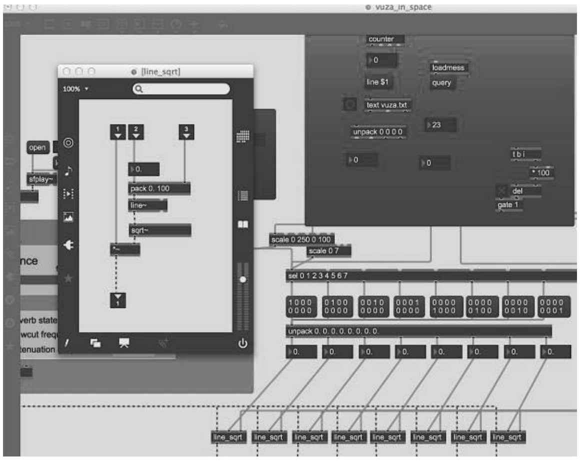

Figure 6. A screenshot from the MaxMSP environment. On the left side the realization of the equal power panning law.

Then, the R and S array elements are scaled, if necessary, and sent to the *line_sqrt* subpatch that realizes the equal panning (as on the left side of figure 6) applied to all the possible transitions between the eight speakers. On the right side of figure 6 it is possible to see how a *sel* object acts to choose the speaker to be activated and how long.

Conclusions

Regular Complementary Canons of Maximal Category were introduced to the musical world in the 1990s. From then on, many algorithms have been developed to try to solve all the problems that such a complex situation causes.

In this paper, we presented the implementation of our algorithm that starts from the Franck Jedrzejewski and the concealed OPEN MUSIC one and explains how to implement and directly manage Vuza canons.

This strategy is very useful especially for a composer's point of view because it allows operating on every stage of the code. Moreover, it is possible to use the so obtained outputs in other software, such as CSOUND, MAXMSP, SUPERCOLLIDER and many others of course, to achieve interesting results. For example, in order to simplify the use of RCCM canons for all users, we are programming a MAXMSP object that returns real-time output of the R and S sets.

In this paper, we showed a simple application of RCCM to create special patterns for real-time sound spatialization. Our research will continue along two lines: study modulations between different Vuza canons and verify the musical relevance of this approach.

Acknowledgments. *Mathematica* fonts by Wolfram Research, Inc.

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[1] A *cyclic group* is a group which should be generated by a single element g, which is called *generator*. This group is an abelian one.

[2] If G is a group, and H is a subgroup of G, and g is an element of G, then $gH = \{gh : h$ an element of $H\}$ is the *left coset* of H in G with respect to g, and $Hg = \{hg : h$ an element of $H\}$ is the *right coset* of H in G with respect to g. For abelian groups, left cosets and right cosets are always the same.

[Abstract in Korean | 국문 요약]
알시시엠 카논RCCM Canons: 실시간 공간화 알고리듬

알바 프란체스카 바티스타 / 니콜라 모노폴리

최대 범주의 규칙 보완적 카논Regular Complementary Canons of Maximal Category (RCCM) 혹은 뷰자 카논Vuza canons은 댄 튜더 뷰자Dan Tudor Vuza가 1990년대 피엔엠Perspectives of New Music 저널에 괄목할만한 논문을 발표하면서 음악계에 소개되었다. 그의 독창적인 견해가 알려진 후 이 중대한 개념은 많은 발전과 대중화 과정을 거쳤다.
음악가들은 늘 캐논의 구성에 많은 관심을 가지고, 플랑드르악파인 조스캥 데프레Josquin Desprez의 복합다성음악이나 요한 세바스찬 바하Johann Sebastian Bach의 대위 기법의 특성들은 공식적인 대수학algebraic 용어로 설명된다. 올리비에 메시앙Olivier Messiaen이 아마 처음으로 리듬 카논의 개념을 소개하고 연구한 이론가 겸 작곡가일 것이다. 수학적 관점에서 봤을 때, 리듬 카논의 구성은 다른 20세기 작곡가들처럼 메시앙도 몰랐을 개념인 두 하위집합의 합으로서 유한하게 순환하는 그룹들의 인수분해로 공식화될 수 있을 것이다. 요즈음 모자이크 카논mosaic canons을 만드는 과정에는 프로그래밍 언어가 적용될 수 있다: 작곡가들이 창조적 음악 작품을 지을 때 형식적 시스템formal architectures의 기초로 이를 활용하여 극도로 복잡하고 거대한 구조를 다룰 수 있다. 이 글에서는 프랑크 예드르제브스키Franck Jedrzejewski의 것에서 비롯된, 작곡가가 알시시엠 카논을 만들고 직접 다룰 수 있게 해주는 알고리듬으로 울프람 매스마티카Wolfram Mathematica에서 저자들이 시행한 것을 보여주고자 한다. 덧붙여 이 알고리듬은 실시간 공간화를 위한 특정 패턴을 구성하는 데 사용된다.

Locating and Utilising Inherent Qualities in an Expanded Sound Palette for Solo Flute

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In the search for an idiosyncratic improvisatory language of solo flute performance, it is sometimes necessary to move beyond the scope of traditional and extended techniques into the world of instrument-extension through computers. To this end, Alice Bennett's creative work leads to the exploration of new electroacoustic techniques, searching for ways to expand the available sonic palette. This paper documents an electronically-extended flute performance utilising Vincent Giles' Spectral Domain Microsound Amplification Software (SDMAS) in real-time. The SDMAS amplifies soft sounds relative to loud, by real-time input, shifting the partials by amplitude around an amplitude-based pivot point. The result is that these otherwise-inaudible partials increased to audibility alongside the higher amplitude partials, drastically altering the perceived timbre of the instrument or instruments being treated. This allows the performer/composer to not only discover, but to exploit a greater range of timbres than are available to a purely acoustic instrument. These explorations lead to a sonically enriched and idiosyncratic improvisatory language based on this hyper-instrument configuration.

Uncovering additional sonic worlds that are inherent in acoustic instrument sound production has been the goal of composers and performers for decades now, possibly as far back as Schoenberg's *klangfarbenmelodie* (Schoenberg 1978) and certainly being well explored and documented by Lachenmann with his *musique concrete instrumentale* (Lachenmann 2004). It has been a feature in the jazz and improvisational world too for quite some time, particularly coming out of the tradition of free jazz, with composer-performers like Ornette Coleman (Ornette Coleman Double et al. 1998) and Derek Bailey (Bailey 1993) exploring the expanded sound palettes offered through extended technique on their instruments.

In parallel to these activities, computer music has facilitated the synthesis, analysis, and resynthesis of sounds that, 120 years ago, were barely dreamed about. Composers such as Stockhausen (Stockhausen 1968), Berio (Berio 1971), Boulez (Boulez 2010), and many others have sought to capture the idiosyncrasies of electronics and electronic processing, at times augmenting an acoustic instrument through electronic means, or a whole ensemble through spatialization.

Now in the popular musical world, effects pedals (electronic augmentation) are a common feature, and digital audio workstations (DAWs) allow for sound manipulation that once took massive effort when working with tape to be done quickly and with far less effort. So we come to somewhat of an impasse: the extraordinary expansion of acoustic sound production using instruments throughout much of the latter half of the 20th century and into the 21st century, and computing power and sonic exploration that continues to uncover new methods. The realm of microsound explored by Roads (Roads 2001) and others continues to be of interest to many composers and performers. Roads' microsound is expanded to include the amplitude domain (Giles 2013-2016). The goal of this paper is to explain and explore a novel approach to microsound and instrument augmentation using instruments, merging the practice of composer Vincent Giles and flutist/composer Alice Bennett. Giles' practice encompasses computer music, notated instrumental music, and installation work. His programming has come about by necessity, rather than from a background in programming, and so gives rise to idiosyncrasies in technique and realisation that a coder may not come across. Bennett is a classically trained flutist who specialises in contemporary repertoire, including an improvisational practice based on the electronic augmentation of the bass and concert flutes.

Spectral Domain Microsound Amplification Software (SDMAS)

The purpose of this software is to allow for the FFT-driven amplification of amplitude-based microsound in both real-time (RT) and non-realtime (NRT) situations, resulting in the ability to 'peer within' a sound's spectral content for those qualities of sound that exist at the edges of our perception, but nonetheless have a profound impact on our understanding of sound.

The result of this amplification process can be analytical, as was first intended, or as a device for real-time augmentation of an acoustic instrument that allows those spectral sounds that are otherwise inaudible in the performance of an instrument to become audible. Spectral-domain processing has an important distinction when compared to time-domain processes (such as compression and expansion) that may or may not achieve similar results, and that is that time-domain processing is frequency-agnostic, dealing only with the sum of frequencies and amplitudes at any given sample point[1]. This frequency-agnosticism makes time-domain processing apply only an increase or decrease in amplitude of the summed frequency and amplitude data, rather than specific frequency or amplitude-based manipulation, which is only possible in the spectral domain.

The remainder of this section of the paper will describe each process within the SDMAS, detailing the algorithms and processing that are employed, and finally considerations for future development and refinement.

Stages of Signal Processing

Figure 1 (below) shows the flow of signal and derived data throughout the patch. This can be expressed in text in the following way:

Audio: Input (RT or NRT) --- Filter 1 --- FFT --- Filter 2 --- Output Stage
--- Output Stage

Data: Input --- Frequency and Amplitude Approximation --- Filters
--- FFT Control

Figure 1. Data-flow text

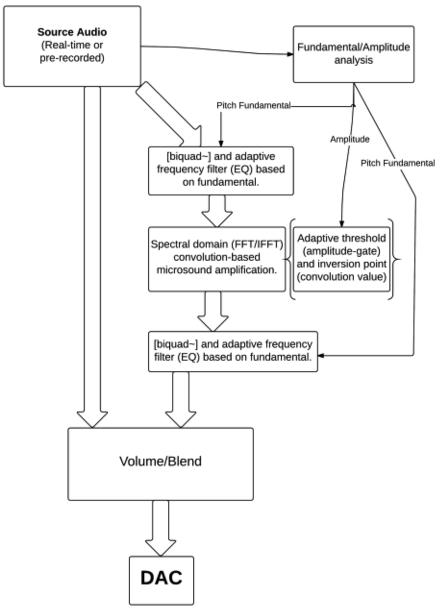

Figure 2. Signal and data-flow diagram of SDMAS.

The flow of signal and data corresponds in part to the layout of the interface for the patch. What follows is a description of the role of each interface component, with sub-sections for technical information, focusing on the algorithms inside the FFT sub-patch. The inner workings of the SDMAS are rudimentary, and the mechanism for amplifying microsound is equally simplistic. The most complex component is actually restraining the amplification system to avoid all of the background white noise inherent in any audio signal, whether RT or NRT.

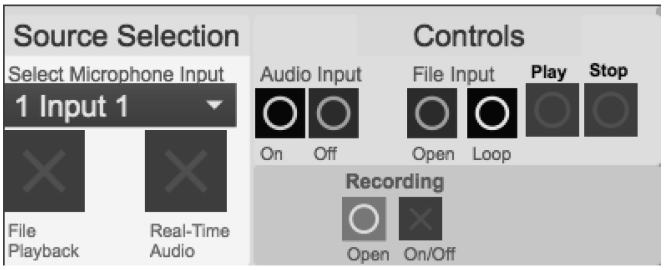

Figure 3. Source selection and playback controls

Figure 3 (above) correlates to the top left box in Figure 1, and has the following functions.

Source Selection

Within this sub-section a user can select the input from available inputs on their audio interface, and route either the file playback or incoming audio through the SDMAS system.

Controls

The audio input buttons switch the [adc~] (hardware input) object on and off, while the file input options include open: open an audio file; loop: loop it; play & stop. As of version 0.9, this is rudimentary playback from disk, not from a buffer. The recording stage just allows for the recording of SDMAS output to disk for later use.

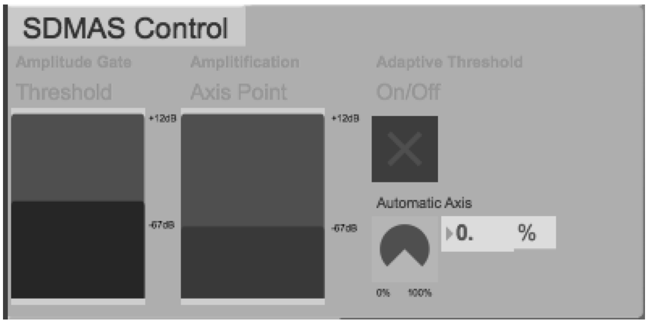

Figure 4. SDMAS controls

The primary user input controls for the FFT-based component of the software consist of a manually controllable amplitude-based noise gate, an axis point (both in decibels ranging from -90dB to ~+12dB), and control for automated threshold and axis point generation based on the incoming signal.

The amplitude gate threshold slider sends the output to the FFT sub-patch as an amplitude value between 0 and 1, all sound below that amplitude will be discarded from the signal path. The axis point needs to be set somewhere above the threshold, because there will be nothing to act on if it is below the value set by the threshold. To make this easier, the automatic axis dial allows the user to set a percentage above the threshold for the axis point to be automatically set to. The adaptive threshold toggle simple takes the peak amplitude of the incoming signal and uses that to modulate the threshold slider.

Inside the FFT

Threshold and axis point functions within the FFT can be described with the following pseudo-code statements:

IF
$([a]mplitude < [t]reshold) discard$
$ELSE\ pass$
$THEN\ FOR\ ((a > t)1/a)$

This process is simply a means of removing excess sonic information (background noise) and inverting the frequency and amplitude information around a given point, using *1/x*. The result is that around any given axis point, all amplitudes will be inverted with soft becoming loud, loud becoming soft. However, the frequencies to which those amplitudes belong *do not change*, meaning that the frequencies themselves become more audible.

Finally, this inverted signal is sent to a convolution[2] system to reinforce dominant frequencies. Figure 5 (below) shows the patching for this process within a [pfft~] sub-patched FFT, running at 4096 window size, using a hamming windowing function, with an overlap factor of 8.

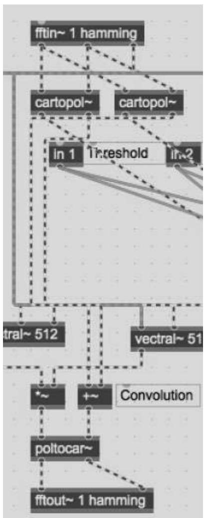

Figure 5. FFT-based convolution

Filtering

Filtering occurs in two stages: before the [pfft~] sub-patch, and after it. It is the same filter on both sides, and can be controlled by fundamental frequency estimation automatically, or controlled manually. The default filter is a bandpass, which results in trimming the excess sound going into and out of the FFT. Wider bandwidth (Q value) allow for greater upper partials, however, narrow bandwidth allows for a tightly controlled emphasis on certain frequencies that may be of interest. This is particularly useful when the user wishes to enhance subtones or very high frequency overtones, and having the filter before and after the FFT cleans the sound up substantially. Figure 6 (below) shows the filter graph and other controls for this component of the SDMAS.

In the top-right corner of Figure 6 is the threshold control for the [fzero~] object which estimates the fundamental and peak amplitude of a given analysis window; the threshold control allows the user to specify the lowest amplitude above which new pitches and amplitudes will be reported. The second option in the top-right corner allows a user to change the type of filter being employed.

Figure 6. Signal cleanup and filtering

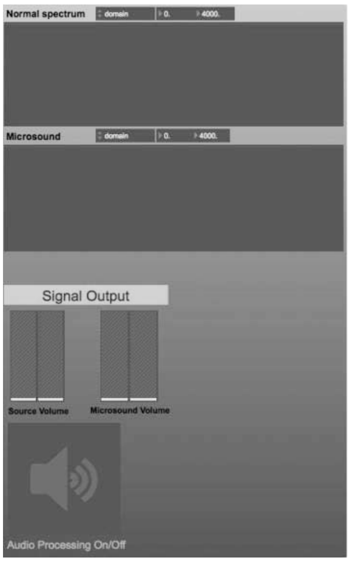

Figure 7. Spectral analysis and signal mixing

Output and real-time analysis

Figure 7 (above) gives the user a real-time spectral analysis in the form of a spectrogram of the source audio and the microsonic components that have been enhanced through the use of the SDMAS patch. The final stage is the mixing between source audio and microsound, allowing the user to blend the two signals to their needs.

Considerations and Future Development

Real-time processing, as described elsewhere in this paper, can be problematic even though the results are extraordinarily sonically interesting. Nonetheless, the system works to highlight aspects of the amplitude-domain in an incoming signal that are otherwise inaudible, contributing to a sense of a hyper-instrument that is not augmented through additional registers or new synthetic sounds, but takes advantage of the inherent sound features of the instrument and combines those with the conventional sound, creating the hyper-instrument.

Non real-time processing does not have the control problems that plague the RT context, so the system is being deployed as part of an analysis methodology to uncover microsonic qualities within works already composed for acoustic instruments, along with acousmatic sounds. The system provides a novel approach to generating new material in an acousmatic setting while also providing a functional tool for analysis and performance.

As of the end of August 2015, the project files have been placed on GitHub[3] and Giles invites contributions to the code in an effort to overcome the problems outlined herein. In the future, the project could be re-written using SuperCollider or similar text-based programming environment to allow more precise control over the algorithms and more efficient use of processing power.

SDMAS and acoustic instruments: why?

This section of the paper will deal with the use of the SDMAS, particularly in relation to the Bennett's artistic practice: improvising and composing with solo bass and concert flutes. In particular it examines how the SDMAS can be used from a creative perspective to augment the inherent sounds produced by these instruments in an improvisatory language and in the composition *Wilf's Lament* (Bennett 2014).

Amplification of flutes, especially softer techniques such as whistle tones and closed-hole technique, is widely used in contemporary performance practice, in notated art-music, jazz, and other contemporary improvisation (post-classical). Simple amplification can be thought of as using a microphone and loudspeaker, analogous to an electric guitar or singer, and allows these techniques to be heard more easily in situations such as: concert hall ensemble performance, open-air outdoor venues, and performances with larger audiences which without amplification would only be audible in intimate concert settings with solo instruments. Examples of this type of amplification include: Paul Méfano's *Traits Suspendus* for amplified contrabass flute (Méfano 1980), Matthias Ziegler's album *Uakti* (Ziegler 1999) featuring amplified low flutes, and Mary Finsterer's *Ether* for solo flute (Finsterer 2001) featuring amplified whistle tones. Jean Penny (2009) discusses the effect of amplification on flute techniques such as whistle tones and breath noises, which she refers to as microsounds, and how these microsounds extend the creative possibilities of the flute. The sounds under discussion in this paper are quieter than those of whistle tones, and are intrinsically linked to the production of sound on a flute, for example: partials, mechanical sounds, and mouth noises that usually would only be audible by the performer. Therefore the purpose of the SDMAS in this performance and compositional context is to go further than simple amplification by exposing this microscopic sound world inherent in the physical action of producing sound with a flute.

Discovering the SDMAS and its uses for augmentation

From a performance and improvisation perspective, it is important to understand how a system (or instrument) works in order to use and exploit it. What follows is an account of a performer's interaction with the software in an attempt to determine the exact effects of the variables in this idiosyncratic augmented performance system.

When learning a new software interface it seems prudent to just experiment. A good place to start with the SDMAS is to turn the device on and observe the decibel and frequency level of the background noise in the room, providing a baseline from which to approximate settings that may be useful. Next, it is best with the SDMAS to test the various techniques you would like to use, in order to again establish a baseline from which to operate. While it is possible to do this in real-time, it is helpful to instead use pre-recorded examples to avoid having to constantly stop playing to make notes and try new settings. Instead, the settings can be manipulated in real-time with an immediate sonic result. This process of exploration allows, with some practice, for a performer to intimately understand the system and attempt to counteract and work with idiosyncratic behaviour of the software itself, resulting in an idiosyncratic language that exploits the inherent qualities of the flute.

Describing the relationship between flute technique and SDMAS augmentation

Each technique on the flute produces different results with the SDMAS and may require changes in settings. This section discusses Bennett's explorations of flute technique and how the SDMAS modifies that signal, and what settings within the patch are most functional.

Whistle tones: the SDMAS clearly enhances the fundamental frequency and the overtones of this technique along with any breath or mechanical sounds that are incidentally produced. The effect is similar to the simple amplification of whistle tones discussed above, but stronger and richer, with a greater range of frequencies added to the sound through amplification. The most effective settings for whistle tones are:

- threshold: -24dB;
- axis point: 25% above the threshold;
- wide frequency bandpass at 1000Hz and unity amplitude, and;
- fundamental tracking switched on for more precision and less partials.

Closed tone hole techniques: the SDMAS amplifies lots of partials creating a sound that is indistinct. This technique results in a wide variety of microsounds that are dependent upon what physical point of the instrument the microphone is attached or pointing at. Amplifying the body of the flute result in more high-end frequencies being amplified, whereas amplifying the head joint results in lower frequencies. The addition of fundamental tracking results in a more distinct sound, and mixing both the source and microsound signals into the output produces the most detail. Other settings are the same as with the whistle tones mentioned above.

Aeolian/residual tone: for this technique it is best to have a wide bandpass without fundamental tracking due to the wide range of frequencies produced. The produced effect is not as complimentary to the aeolian tone as it is to other techniques as the SDMAS produces what could be described as filtered white noise, rather than augmenting already existing timbral qualities. As above, all other settings are the same.

Key clicks: the SDMAS amplifies the pitch of the key clicks and accentuates difference tones produced by trilling the clicks. Placing a microphone at the head joint gives more pitched material than amplifying at the keys. Tapping the different sections of the flute with fingernails is also very effective as the SDMAS amplifies the sounds resonating from the head joint, like a percussion instrument. The most effective settings are:

- threshold: -24dB;
- axis point: -12dB, and;
- narrow frequency bandpass at 1000Hz and unity amplitude.

Other settings are the same as above.

Soft sounds (*ppp* to *mp*): because these 'soft' sounds are by comparison loud, the best settings are:

- threshold: -12dB;
- axis point: -8dB, and;
- narrow frequency bandpass at 1000Hz and unity amplitude.

The effect on these pure tones (non extended techniques) is the amplification of clear overtones with some breath and mechanical noises. It is both effective and highly expressive in real-time performance.

Louder sounds (*mf* to *fff*): due to the amount of sonic information contained above the amplitude threshold at this dynamic, the SDMAS amplifies a lot of partials, producing an effect that is very distorted. This sound changes and can be shaped with different tone qualities and embouchure shapes. For example, in the low register (C4 to F4), a warm, open tone will result in a less distorted microsound signal than a hard, edgy tone that has many more amplified partials. While this effect could facilitate an interesting study of tone colour, it is quite overpowering and its use is limited. The most effective settings are:

- threshold: -24dB;
- axis point: -10dB, and;
- wide bandpass at 1000Hz and -6dB.

Multiphonics, and flute and voice: both techniques produce sonically similar to the above when used with the SDMAS. If the flute techniques can be played at a lower dynamic (*ppp-mp*) then the SDMAS is able to focus on only a few overtones producing usable and interesting results. However, at louder dynamic just like with the unaltered sounds above, it produces a distorted signal.

Exploiting the idiosyncrasies in creative practice

Having explored the possibilities of a new sound world with the SDMAS, the next step is to internalise these effects and develop control over them, both with the acoustic instrument (adapting the playing to the different effects of the software) and with the device (controlling the settings). There are two ways to go about that the latter: leaving the SDMAS settings in place or having interactive control over them. The first option is much simpler and can have very effective results, however the performer is limited to certain techniques, due to the incompatibility between different techniques within the

SDMAS; such as whistle tones and conventional timbres played at *forte*. This clearly limits the expressive potential of the instrument and the SDMAS. *Wilf's Lament*[4] (Bennett 2014) was composed with this in mind, exploring the quieter sounds of the flute and using the SDMAS to amplify the minute detail of whistle tones, harmonics and difference tones, rather than using louder, pure sounds which risk SDMAS distortion and loss of control.

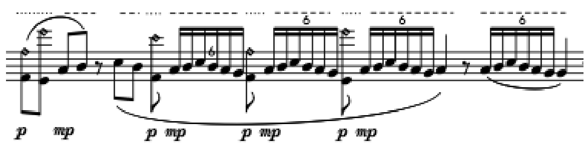

Figure 8. Excerpt from Wilf's Lament (Bennett 2014)

Interacting with the SDMAS directly and in real-time would be the most functional way to perform highly interactive pieces and utilise this newly uncovered sound palette. This could be achieved either by collaborating with a sound technologist during performance (such as in Time and Motion Study II (Ferneyhough 1978))or by the flutist using a foot pedal. As with any ensemble performance, communication is key and the performer must be very clear with the sound technologist about what kind of effects they are looking for, desired settings, and so on. This arrangement is suited to written music where the technologist is able to follow a score and make the necessary changes when instructed or cued, but may not be as useful in improvised performance.

In improvisation, a foot pedal can give complete control of the software to the performer, though this requires a considerable amount of preparation and coordination between hands and feet, which is not usually part of a flutist's performance practice. To facilitate this, a floor-based MIDI control pedal must be programmed into the Max patch, with the required pedals being mapped to pre-set functions that the performer can then employ as needed. An example might be having the settings most conducive to a whistle-tone being mapped to button number 1, while the settings best suited to key clicks could be mapped to button number 2, and so on. A foot pedal could be mapped to control settings based on the pre-set, such as threshold, axis, EQ-width, and the mixer settings, enabling a performer to adapt the settings to suit the technique they are employing in real-time, and to exclude any unwanted background noise generated by the FFT or the room.

The performer must practice using the controls so that they are comfortable with them as they are with their acoustic instrument and able to move their feet without detracting from the performance.

Concluding Remarks

This paper has explored the technical workings of the SDMAS system, and an approach to utilising it in a live performance context. It uncovers some problematic areas and suggests future development to the system, along with methods of circumvention or adaptation to get the system to work in these situations. The processes described herein offer insight into the sonically rich worlds of *amplitude domain* microsound, as distinguished from the conventionally used *temporal domain* microsound, and offers a novel and flexible system for exploiting those microsounds through FFT in both a real-time and non-realtime contexts for instrument augmentation or analysis. Specifically, we examined how a solo flutist could develop an improvisational and compositional language based upon the idiosyncratic sounds that are inherent in flute performance, dramatically expanding the sonic palette of the instrument and building upon those expansions already mentioned in the introduction, through extended techniques.

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[1] This problem of summed frequency and amplitude data over time is the subject for further research into microsound.

[2] This is the inverse of the complex convolution example in the Max tutorial system: https://docs.cycling74.com/max5/tutorials/msp-tut/mspchapter26.html/

[3] https://github.com/vgiles/SDMAS/

[4] https://abennett.bandcamp.com/track/wilfs-lament/

[Abstract in Korean | 국문 요약]

플루트 독주를 위해 확장된 사운드 팔레트Sound Palette의 고유가치 탐색과 활용

알리스 베넷 / 빈센트 길레스

플루트 독주 특유의 즉흥적 음악언어를 추구하면서, 전통적이고 현대적인 연주기법의 범위를 넘어 컴퓨터를 이용한 악기의 확장이라는 영역으로 넘어가야 할 필요성을 종종 느낄 때가 있다. 이러한 이유로 알리스 베넷Alice Bennett의 창작 작품은 실용적인 사운드 팔레트를 확장시키는 새로운 전자음향적 기술을 탐색케 한다. 이 글은 빈센트 길레스Vincent Giles가 실시간으로 분광 영역 마이크로사운드 증폭 소프트웨어SDMAS(Spectral Domain Microsound Amplification Software)를 사용한 플루트 연주에 관하여 기술한다. 분광 영역 마이크로사운드 증폭 소프트웨어SDMAS는 실시간으로 입력된 소리가 특정 음량 기준점 주변 배음들의 음량을 변화시켜, 작은 소리를 상대적으로 큰 음량으로 증폭시킨다. 그 결과 들을 수 없었을 [작은] 배음들이 큰 음량의 배음들과 같이 커져, 기존의 악기 혹은 변형된 악기 음색과도 확연히 다른 음색을 얻게 된다. 이는 연주자/작곡가가 순전히 어쿠스틱 악기에서 얻을 수 있는 것과 다른 음색을 보여줄 뿐 아니라 보다 넓은 영역의 음색을 활용할 수 있게 해준다. 이는 초월적 악기의 확장형태instrument configuration에 기반한, 음향적으로 풍부하고 독특한 즉흥적 음악언어를 이끌어 낼 것이다.

The Immersive Lab: An interactive audio-visual space for artistic exploration

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The Immersive Lab is an artistic and technological research project of the Institute of Computer Music and Sound Technology at the Zurich University of the Arts. Participating artists create works for a media space, which integrates panoramic video, surround audio with full touch interaction. The main challenge for the artists is to find ways of translating touch into interaction models for their audio visual ideas. At the end of the process, works are presented to the public in the form of interactive audio visual compositions. This article exposes the technical structure of the installation, describes the underlying concepts of the project, and addresses the diversity of artistic ideas in the thematic, conceptual as well technical domains. Example works by participating artists are discussed and show in what way the central topics of the Immersive Lab installation revolve around observation, physical reaction, individual and group interaction. The diversity of ideas and the considerable number of completed artistic works confirm that the advantages of this project, i.e., the integration of interactive audio visual modes in one media environment, indeed inspire practical, spontaneous, and collaborative learning and experience.

> Yet immersion viewed from a human standpoint is achieved once our physiological, creative, intuitive, intellectual, and imaginative functions are engaged and transcend basic disruptions and distractions. This immersion is not the escape offered by cinema or by the CAVE environment for 3D computer graphics, but is a multifaceted and active human experience of connection and communication that involves both internal and external dimensions. (Kozel 2007: 145)

The Immersive Lab is an artistic and technological research project of the Institute of Computer Music and Sound Technology at the Zurich University of the Arts. It is a media space that integrates panoramic video, surround audio with full touch interaction on the entire screen surface. The Immersive Lab provides a platform for artistic works that are tailored to the unique situation that this configuration offers. These works articulate the relationships between immersive audio and visual media and direct interaction. The lab functions both as a space for experimental learning and creation and as an audiovisual installation for the public, showing the finished pieces in a self-explanatory way inviting exploration.

Immersion

Immersion in its original sense means being submerged or enveloped, usually in water. In media arts and theory this term has been extended to mean envelopment by mediated contents, be they visual, sonic or sometimes even tactile. Older forms of mediated immersion can be frescoes set in architectural spaces (Almond 2011) and panoramic paintings (Grau 1995: 62). An important concept in the discourse about immersion is 'virtuality' (Nechvatal 1999), the idea that mediated contents generate an artificial envelopment. Here the embodied perspective (Varela et al. 1991) informs the perception of the digital image itself, it becomes integrated into the body's process of perceiving it (Hansen 2004). Cinema has been for a long time the principal vector for immersive experiences for a large public (Rose 2012) and pushes further into that domain by the application of 3D and stereoscopic techniques (Zone 2007). Video games in general and the recent resurgence of virtual reality headsets have become another important way of experiencing virtuality in an actively engaged manner (Ihde 2002: 81).

In contrast, the Immersive Lab seeks a different blend between the virtual and the 'actual' (Schechner 1997) by giving the audience a direct physical engagement with the media within the installation's space as well as providing the occasion for social and group interactions. In the entire project's scope, the communal aspect effects not just the visitors. Working in the Immersive Lab also presents the necessity for collaboration and sharing of experience, techniques and knowledge. Working in this way raises a number of questions:

- In which way can the notion of immersion inform artistic creation?
- To what extent can attention be shaped, in a situation that presents its contents and interaction in a surrounding format?
- Where are the boundaries between personal and collective observation?
- With which elements does a work have to be composed

in order to take the multiple modalities into account?

- On what level does immediate tactile interaction transform the experience of engaging with digital art-forms?
- How can interactivity transform the relationship between abstract creation processes and sensorial experience?

Project Concepts

The Immersive Lab installation as a platform is the fruit of several years of investigation, development and artistic creation. The term Immersion is used in a broader sense. Apart from spatial envelopment by image and sound, additional levels of immersion are generated for the visitors: they enter into a dedicated physical space, the direct tactile interaction on the panoramic surface enhances their personal engagement, and finally within the shared space arise group behaviors and social interactions. Such an extended form of immersion provides a multi-faceted experience.

The artistic works developed for this installation can be collaboratively created and combine visual and sonic material with generative and algorithmic methods. The focus of the installation is on artistic approaches of creating real-time pieces that react to visitor interaction and that take advantage of the panoramic nature of the installation.

Work in the Immersive Lab happens in different phases and activities, and addresses different people. In a teaching context, in general, any student can visit the lab in guided tours. Students majoring in electronic music or media arts, however, are invited to actively learn by exploring the inner workings of existing pieces. Artists and advanced students have the opportunity to become involved more intensely by creating new pieces. For this, the ICST offers to share its experience, methods, and tools for development and realization of ideas for this particular media space. Finally, in the exhibition context, general audiences are invited to experience the entire catalogue of works.

Different forms of engagement are possible within the installation. The audience can freely explore the works and experience different types of perceptions. Artists can experiment with the development of compositional strategies for working with different senses and artistic domains. Thus the installation exposes foundational aspects of immersion such as spatial and multi-sensory perception, which provide relevant topics for investigation.

Technical structure

The basic construction of the Immersive Lab consists of four sections made of free-standing metal frames that are covered with screens. Together these four panels form a cylinder of approximately 4 meters diameter, with a screen surface that spans 10.3 meters by 1.5 meters. By using four projectors to cover the cylindrical canvas an image is produced, which currently has a resolution of 5120 x 720 pixels (four times HD at 720p). Placed outside the structure at a certain distance behind the screens are four mirrors that divide by two the distance required for image projections as well as capturing tactile information. With this disposition, the minimum area required for setting up the installation structure is seven by seven meters, not counting the access and technical working areas (see Figure 1).

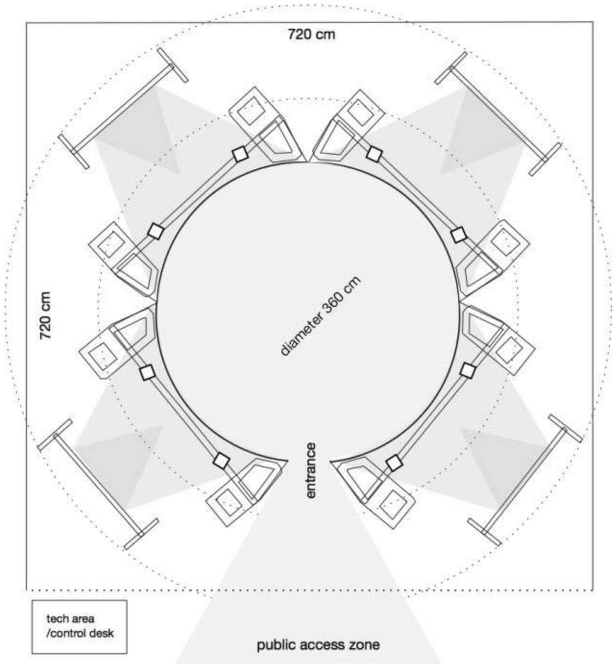

Figure 1. Floor-plan of the Immersive Lab

Touch interaction

Interaction with the installation occurs by touching the screens with the hands. The resulting points are measured by tracking software that is observing the screens from behind in infrared light. The complete system consists of infrared lighting on the edges of each canvas, cameras equipped with infrared filters that capture the visual field of each screen and software on dedicated small computers that process the signal from the cameras. The software applies standard OpenCV[1] computer vision algorithms and is capable of detecting a large number of points. Due to the delicate illumination situation within this circular arrangement, the detection reacts to fingertips but also elbows, shoulders, backs and other body-part that touch the canvas. The information obtained by the four sub-systems is processed locally and dispatched to a master application through OSC[2] to main computer. The touch information is transmitted via the TUIO[3] protocol (see Figure 2).

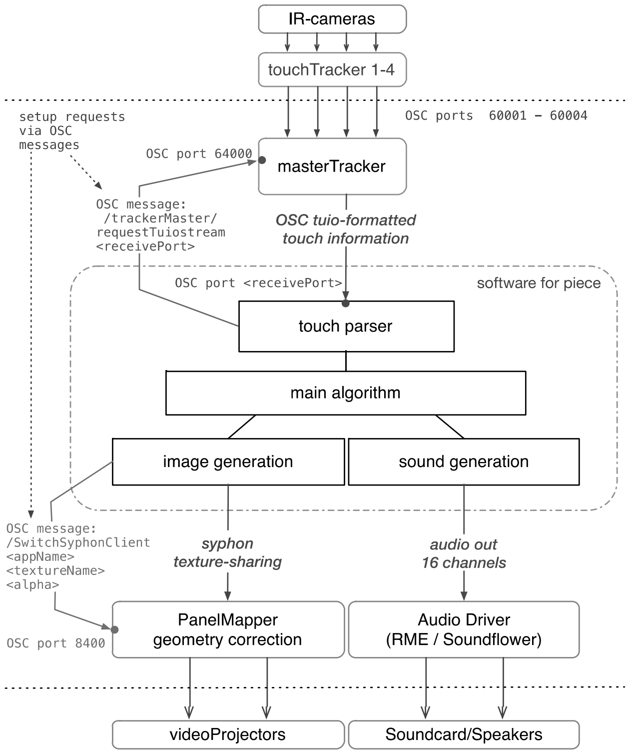

Figure 2. Software flowchart, connection and protocols

Image geometry correction

Distortion of the image presents an issue that needs to be solved when projecting from the outside onto a cylindrical screen. In the Immersive Lab system the counter-distortion for straightening out the image on the cylinder is done by a dedicated standalone application, which receives the output image from each piece's graphical software via a shared texture-bus on the graphics card called 'Syphon'[4]. In addition to geometry correction this application also offers the possibility of crossfading between images from two applications.

Audio system

The audio system consists of 16 speakers uniformly distributed on two levels as well as two subwoofers. The speakers are located behind each of the four screens and are arranged at the corners of the frames, thus avoiding the projection beam and forming two concentric circles (see Figure 3).

Platform and Programming languages

Artists creating work for the Immersive Lab may use a variety of programming languages on the Mac OS platform. One of the main objectives of the project has been to enable working on many different types of programming languages. Commonly used software environments are MaxMSP/Jitter, Supercollider, Processing and OpenFrameworks. For participants the main challenge so far has been to find ways of interpreting the touch information received into meaningful interaction models for their audio-visual compositions.

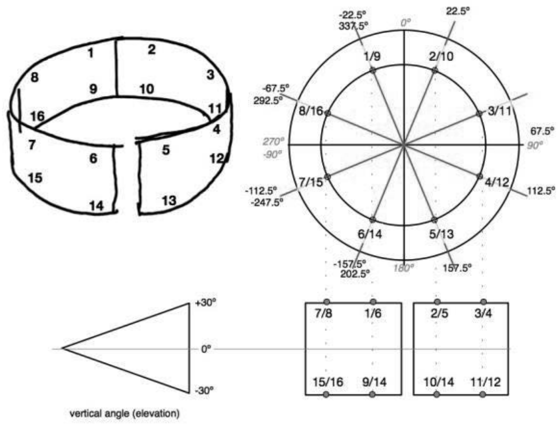

Figure 3. Speaker arrangements

Works

In the past three years the Immersive Lab project has seen the participation of a number of artists from the fields of music composition, media arts, creative coding and interaction design. The works are different in many of their aspects, which implies that the project offers opportunities for exploring ideas from different perspectives.

By following the interviews with the participating artists, it becomes possible to understand the diversity of ideas in the thematic, conceptual as well technical sense. The following four examples show the range of approaches present in the current collection of works.

Figure 4. Andre Sier – Hyperborea

In his work '*Hyperborea'*, Andre Sier, a media artist from Lisbon, Portugal, invites the audience to collaboratively interact with an abstract 3D-world through a video-game logic. Successful cooperation between the audience is rewarded with a transition to the next level. He states that his piece is:

> an interactive immersive audio-visual environment forging the experience of travelling towards and inhabiting Hyperborea, inspired by ancient Greek myths, where in the land beyond the northern winds region, overseen by Boreas' ruling and dynamics, through a long and perilous journey, far far away, lays Hyperborea ... This work provides audience the machine code infrastructure to walk this windy path to Hyperborea, through a game-like system architecture with 100 levels and haptic interaction, aural and visual spatialization. This vehicle opens its windows and offers audience the control. It' s up to the audience to reach it, touch it, or to get lost along the way. The audience may engage on the path to Hyperborea individually, or collaboratively, ... Many have tried to taste its soil, feel its texture, rarely finding their destination. (Sier 2015)

Figure 5. Jan C. Schacher – Clocks and Clouds

Jan Schacher invites the audience in *'Clocks and Clouds'* into a simple mode of interaction: each visitor creates her own audio visual impulses. In this way, the installation obtains the role of an orchestra where each visitor plays their own instrument and through collaboration creates a unique whole. The core idea is that:

> In the piece 'Clocks and Clouds', the circular space is seen as a cyclical movement space that is filled with acoustic and visual pulse-trains of differing intervals and life spans, which are triggered by the user's touch. The combination and interaction by several users generates a complex web of overlaid visual and acoustical 'voices', which merge in perception into groups or masses. … The main motivation for this piece is a curiosity about perception of temporal, spatial and sonic densities. The piece serves as an investigation into how our perception is capable of separating visual and acoustic streams and how sensory overloading has the effect of forming groups or combined objects or gestalts that exist in perception. Another topic of interest is the social interaction by several visitors, in particular in relation to such a reduced sonic and graphical space. The stark or abstract quality poses a challenge to perception. (Schacher 2015)

Figure 6. Terence McDermott – Star Chamber

Australian media artist and sound designer Terence McDermott created a piece where rich audio visual content is generated by a mathematical algorithm. The interaction is indirect, affecting only state changes of the piece. In this way, the audience is forced into deep thinking about the piece and observation of other participants' actions.

He describes his piece '*Star Chamber*' as:

> An exercise in the sonification and visualization of data. Chaotic data is generated by a simple mathematical algorithm, and it is represented as particles which emerge and die away in a virtual space. These same particles also have a sonic presence, to articulate the structures generated. It is a representation of an abstract mathematical space within a real space (the listening/ viewing space), mediated through a virtual space. In this sense, the piece can be seen as a spatial collage, with disparate spaces layered upon each other. The idea of collage is further extended by juxtaposing the representational (data visualization) with the expressive (human voice). Here, the piece attempts to enter another space, the so-called psychological space. Representation of abstract structures is usually arbitrary, and based on principles of convenience and clarity. The mode of representation itself does not contain any expressive meaning. This piece attempts to play around with the idea of meaning and representation, by deconstructing the voice, through its particularization, separating out its symbolic meaning from the sounds themselves, taking us back in time, to a preverbal state. The title refers to a medieval court of law where, if individuals were seen to behave immorally, but not technically breaking any law, they could be summoned by the Council of the Star Chamber to defend their actions and reflect on their past deeds. (McDermott 2015)

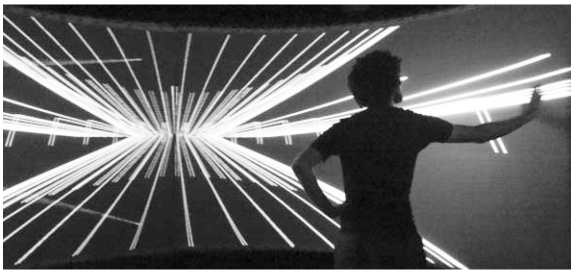

Figure 7. Ted Davis – FORMBIT

American media artist Ted Davis created a work '*FORMBIT'* with an almost dizzying graphical impact. Shapes drawn by the visitors enter into slowly rotating and phasing movements and generate sound with a hypnotic effect.

He states that his work:

> Revolves around interactively positioned nodes, forming the 2D basis for 3D explorations. Through rotation and transposition, each structure becomes a 3D extrusion of itself, immersing the visitors as it orbits at oscillating speeds around them. This creates a feedback loop for the visitors as they design within the Immersive Lab's multi-touch surrounding and react upon the generative output. With each new structure replacing the older of two FORMBITs at any given time, an array of Moiré effects can be experienced through the lines' subtle offsets and overlaps. Rather than reacting to audio, this work investigates generating audio from the visuals. This is accomplished by measuring the luminosity of light passing each of the 16 audio channels' physical zones, which are mapped across a virtual synthesizer's frequency and amplitude. Additionally, the speed of rotation and distance of orbiting forms are passed to the synth, modulating and phasing these frequencies in an intensity that reinforces the visuals being projected. The minimal aesthetics and interaction of this work provides the visitor with a clear understanding of their own interactions within the Immersive Lab and an engulfing cinematic experience when standing back to enjoy how their FORMBITs evolve over time. (Davis 2015)

Discussion

With these quotes from the pieces' program notes it should become evident that the works are thematically and technically diverse and represent different approaches. A variety of techniques is used to generate audio-visual processes, and different methods engender specific interactive worlds. Each of these works opens up a specific perspective onto the field of interaction and immersivity.

Some of the artists use the Immersive lab space to present scientific work through visualization and sonification. Here the role of observation rather than the interaction becomes more dominant. Other artists such as Ted Davis combine the role of interaction and observation. The foundation of many of the works within the Immersive Lab are generative processes. Thanks to this technique the installation space can be active even in the absence of an interacting audience. The artistic freedom in defining interaction through algorithms can be considered as one of the specific strong points of this installation.

These example works demonstrate how the basic mode of engagement for the audience in the Immersive Lab installation are observation, physical reaction, individual and group interaction. These behaviors are necessary in all the works but in different proportions. Contrary to similar installations, such as UCSD Qualcomm Institute's StarCAVE Cave system (DeFanti et al. 2009), the Recombinant Media Lab's Cinechamber (Vikram 2014) or the installations using the Extended View from Graz (Venus et al. 2011), the Immersive Lab promotes collaborative, generative, interactive audio visual compositions. Each Immersive Lab piece begins its life with an original idea by the artist and is developed through cooperation with the members of the project and contact with the audience. This type of cooperation is the key factor that contributes to the successful artistic realization of a number of different ideas as well as efficient and spontaneous learning and experience for the artists and audience.

Future work

In the future the Immersive Lab project expands its reach by engaging with students and artists from different countries and the new forms and ideas they bring. This process imply changes to the installation structure. The most common suggestions are related to using additional devices, either temporarily or permanently integrated within installation's structure. One of the ideas is to introduce of a depth-sensing camera such as a Kinect. In this way the interactive zone could be expanded from a 2D to a 3D space.

Empirical experiments are planned with 'Audicor'[5], a portable acoustic cardiograph. This would represent a first attempt at extending the project to include experiments with physiological data of the visitors.

A further task will be the creation of a database of different workflows and toolsets developed for the Immersive Lab. During the development of their pieces participants create code solutions for various programming environments. Collecting them as examples in a common repository will reduce the time required for future participants to solve technical but also conceptual issues such as touch interpretation. This would give participants more time for their artistic work.

A first version of an Immersive Lab software simulator was completed that allows participants to experiment and develop works on their own computers outside the actual installation structure. The simulator cannot entirely replace the Immersive Lab infrastructure, because of the nature of real-time interaction with multiple hands by an audience. Only the actual conditions in the installation can provide the full immersive and interactive experience. However, through further development and refinement of the simulator software, the time needed for experimentation within the real infrastructure may be reduced and simplify project organization.

Figure 8. The Immersive Lab seen from above (2013)

Conclusion

Works created within the Immersive Lab project represent a blend of art and science set inside an interactive immersive environment. In this context the possibilities of technology, the human response to the digital world and the forms that connect those two worlds can be investigated and experienced. One of the biggest advantages of this project is that it enables artistic experimentation and learning in a hands-on, fully matured media installation. This implies the exploration of different ideas and thematic areas through audio-visual and interactive work and, most importantly, through collaboration between and among the artists and the audience.

Acknowledgments

Thanks to the Institute for Computer Music and Sound Technology ICST at the Zurich University of the Arts, and to all artists who participated in the project for their support and creative work.

The project website: http://immersivelab.zhdk.ch/

The institute's website: http://www.icst.net/

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[1] http://opencv.org/

[2] http://opensoundcontrol.org/

[3] http://www.tuio.org/

[4] http://syphon.v002.info/

[5] http://www.inovise.com/

[Abstract in Korean | 국문 요약]

몰입형 랩The Immersive Lab: 예술적 탐험을 위한 인터랙티브 시청각 공간

보얀 밀로세비치 / 얀 샤체르

몰입형 랩은 취리히 예술대학Zurich University of the Arts 컴퓨터 음악 음향 기술 연구소Institute of Computer Music and Sound Technology의 예술 공학 연구 프로젝트이다. 참여 예술가들은 풀터치 상호작용이 있는 파노라마식의 비디오와 서라운드 음향이 통합된 미디어 공간을 위한 작품을 만든다. 그들의 주요 도전과제는 터치를 통해 시청각적으로 의미있는 상호작용이 가능한 모델을 찾는 것이다. 이 과정의 결과로서 상호작용 시청각 작품들이 대중에게 선보여진다. 이 글은 [몰입형 랩] 설치의 기술적 구성과 이 프로젝트에 내재한 개념을 설명하고, 주제와 관련하여 개념적일 뿐 아니라 기술적인 관점에서 미적 아이디어의 다양성을 역설한다. 예시된 참여 예술가의 작품들이 논의되면서, 어떠한 방식으로 몰입형 랩 설치의 초점이 관찰과 물리적 반응, 개인과 그룹의 상호작용을 중심으로 맞추어지는지를 보여준다. 한 미디어 환경에서 여러 방식의 상호작용 시청각 통합 형태를 이루어내었듯, 이 프로젝트가 실제적이고 자발적인, 협동적인 학습과 경험의 효과가 확실히 있었음을 다양한 아이디어[의 존재]와 상당수의 완성된 예술 작품으로 보여준다.

Audio-Based Visualization of Expressive Body Movement in Music Performance: An Evaluation of Methodology in Three Electroacoustic Compositions

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An increase in collaboration amongst visual artists, performance artists, musicians, and programmers has given rise to the exploration of multimedia performance arts. A methodology for audio-based visualization has been created that integrates the information of sound with the visualization of physical expressions, with the goal of magnifying the expressiveness of the performance. The emphasis is placed on exalting the music by using the audio to affect and enhance the video processing, while the video does not affect the audio at all. In this sense the music is considered to be autonomous of the video. The audio-based visualization can provide the audience with a deeper appreciation of the music. Unique implementations of the methodology have been created for three compositions. A qualitative analysis of each implementation is employed to evaluate both the technological and aesthetic merits for each composition through empirical survey results.

Artists and Musicians constantly evolved the meaning of art and cultivated new forms of art. Not only did they try to harmonize different categories of art, but also attempted to combine technology and art into a new form in the 20th century. In this paper, as a part of the movement, the author introduces the methodology to magnify visual components of music performance, which audience might underappreciate, using audio computation and video equipment. The author reviews the implementation of the methodology, and evaluates its electroacoustic compositions in composer's perspective and audience's perspective. For the objective evaluation, the author discusses the results of the empirical research survey.

Introduction

Expressive Body Movement in Music Performance

Visual aspects of performance can distract from the appreciation of music, and sometimes closing one's eyes can increase the enjoyment of music. However, there are many people prefer to go to a live-concert to appreciate music instead of listening to a high-quality of recording of the same performer. They want not only to hear the sound of music but also to see the expressiveness of the music performance. The former might have a similar philosophic idea of *acousmatic music* in electroacoustic music. In Scruton's perspective, visual aspects of the performance of traditional music can be a hindrance to the reception of its sonic expression (Scruton 1997: 2-3). In what he calls *acousmatic experience,* music ideally exists by itself to an audience in terms of how its sounds are made, just as Pythagoras' students listened to their teacher speaking from behind a veil.

Significance of Expressiveness in Music Performance, In Scruton's opinion, the expressiveness of the performer's movements is an obstacle to appreciate the music, and the audience can receive the expression of the music without seeing the expressive body movements. Deniz Peters discusses the importance of a performer's expression in electroacoustic music. Peters starts by mentioning the way a listener can hear the musician's body in his or her music. Peters describes three aspects: the presence of the performer's body to the listener in the act of listening, the body's appearance of composer in music, and performative listening. He argues there is a residue of bodily presence in the sounds people hear, both in the activity of sound and on the side of the listening experience. Peters concluded his thesis:

> Despite the often noted 'disruption' of bodily sound-making in music using electronic media, and despite listening attitude and theories that construe listening as disembodied, there is much more bodily presence in what people hear when people hear music than hitherto acknowledged, and that, in terms of aesthetic perception, it may even be omnipresent. (Peters 2012)

In Peters' writing, he aims to convince the reader that expressiveness in music is significant enough to merit appreciation, and that an audience can hear the performer in the music itself.

Furthermore, there is much other research to support the importance of the expressiveness of music performance. In musicology, Davidson suggests that vision can be more informative than sound in the perceiver's understanding of a performer's expressive intentions. She proves this through several related experiments in live music performance. In one of her experiments, she separated three modes of observation: vision alone, sound alone, and sound and vision together. The result revealed not only that vision is a useful source of information about expressive action, but also that a combination of vision and sound more clearly communicates the expressive action (Davidson 1993: 112).

Rodger, Crig, and O'Modhrain examined the significance of visible performance when people rated the level of expertise of performers. In their experiments, musicians and non-musicians rated performances by novice, intermediate and expert clarinetists from a sound recording, a point-light animation of their movement, or both. When they switched the sound recording and visual movement, ratings of novice musicians' music was significantly higher when paired with an expert's movements, although the opposite was not found for an expert's sound presented with a novice's movements (Rodger / Craig / O'Modhrain: 1137-50). This result suggests that musicians and non-musicians perceive the expertise of a performer from a musician's body movements in addition to their sound.

In an article by Friedrich Platz and Reinhard Kopiez, they describe a meta-analysis of several existing experiments that investigate expressiveness in music performance (Platz / Kopiez 2012: 71-83). By analyzing the data from fifteen studies, their article examined the relative contribution of sound, music, and visual signals for effective multimedia performance. Similarly, Bergeron and Lopes examined and compared several studies on expressive body movement. Their discussion of expressive properties in music performance mentions not only sonic properties but also visual properties (Bergeron / Lopes 2009: 2). Their article ends with the composer Robert Schumann's words, which are taken from his book. The same quote can be found in Davidson's articles (Davidson 1993; 1995).

> Within a few seconds tenderness, boldness, exquisiteness, wildness succeed one another; the instrument glows and flashes under the master's hands . . . He must be heard and seen; for if Liszt played behind a screen a great deal of poetry would be lost. (Schumann 1946: 156)

Schumann's words imply not only the importance of the sonic properties of musical expression, but also the significance of the visual properties of expressive body movement on the stage. Schumann is reminding us that there is a great deal of expressiveness in music performance, which might be lost when people listen to audio recordings instead of seeing live performances.

Definition of Expressive Body Movement

In most cases, musical performers use their bodies to interact with their musical instruments when they perform. The role of body movement in musical performance conveys the musical intention of the composer not only through the sonic properties of the music performance, but also through the visual properties of the performance. In other words, performative body movement has two functions: first, it inevitably creates inflections in the sonic properties of resulting tones, and second, its visual properties assist to elevate the audience's appreciation of expressiveness of the music. Several scholars define and categorize performative movements from different perspectives. Among the definitions and categorizations, some were adapted to this study, and re-classified according to two distinct aspects of performance movement: sonic properties and visual properties.

Sonic Property of Body Movement. The sonic properties of performance movement are described using several terminologies. Delalande's categorization of body movements designates effective gesture as those which affect the sonic property, which means the movement actually produces the sound (Delalande 1990). Cadoz uses another term—instrumental gestures—to define such body movements. Instrumental gestures are regarded to be a complementary mode of communication to empty-handed gestures (Cadoz 1988: 1-12), and are characterized as a gesture that is applied to a concrete object with a physical interaction and its phenomena during the interaction. These two terms have different perspectives, but a similar meaning in terms of the sonic significance of the gesture.

Visual Property of Body Movement. On the other hand, as a visual property, musicians also move their bodies in a way that is not directly related to producing musical tones. Body movements that assist to increase the expressiveness of music are often called expressive movements, ancillary gestures, expressive body movement or accompanist gestures. Each term has a slightly different meaning. Ancillary gestures in electronic music include the actions that are not used to produce sound, such as walking, posture adjustments, facial expressions, dance movements, muscular tension, and even motionless signals such as brainwaves (Miranda / Wanderley 2006: 10). In Delalande's definition, accompanist gestures are a body movement such as shoulder and head movements, which support expressiveness but can also be related to producing sound. Among the several definitions, the musicologist, Jane Davidson's terminology is suitable to this study in terms of its visual property. Davidson has conducted significant research projects on movement that does not produce actual sound, and has used several different terms to designate the movements: expressive movements, expressive body movements, and expressive bodily movements. Among the terms, expressive body movement is used throughout this paper.

Psychological Research on Expressive Body Movement

Interest and attention to multimedia, especially for relationships between visual and sonic stimuli, can be found in many psychological studies. For researchers outside the field of psychology, particularly those who are in the fields of music and visual arts, a connection with the scientific approach can be a challenge from both conceptual and practical standpoints (Cohen / Lipscomb / Tan / Kendall 2013). However, since researchers in psychology gather data from humans or animals under controlled conditions for the purpose of addressing specific questions with an empirical experiment, artists and composers, especially those who deal with multimedia, can acquire the psychological rationale behind their arts and music.

Through this empirical approach, psychologists investigated the relationship between visual and music, and proved their assumption about human's perception on multimedia. Before multimedia, they already had researched music and human auditory cognition for a century, which is almost the same length of time as multimedia's history. They have researched not only psychological phenomenon on listening to music, but also human's perception of music based on music theory.

As an extension of this research on music, since the 1980s several psychologists have specifically focused on the Congruence-Association Model (CAM) of multimedia. The history of psychology on CAM emitted not only the research and theoretical outcomes but also an example of the process of discovery, which is mostly based on an empirical approach (Cohen 2013). Since the goal of this study is related to psychological views for the reasoning and purpose of composing with multimedia, the basis can be found in the empirical approaches and experiments of psychology on cross-modal perception in multimedia.

Bolivar, Cohen, Fentress, and Iwamiya investigated the congruency between music and visuals with many different experiments, and they categorized the perceived congruency in two aspects (Iwamiya 2013). One is formal congruency: the matching of auditory and visual temporal structures, which produces a unified perceptual form to auditory and visual information. The other is semantic congruency: the similarity between auditory and visual affective impressions. This similarity allows the listener to tie the sound to the visual event, but allows more freedom for artistic parings of visual and sonic event. Both types of congruency have important roles in the cross-modal interaction between auditory and visual domains. While formal congruency produces a united perceptual form between auditory and visual information, semantic congruency helps to convey the meaning of audiovisual content to the perceiver.

Formal Congruency. Formal congruency is the matching of auditory and visual temporal structures, a phenomenon which has been well documented. Iwamiya, Sugano, and Kouda experimented on perceived congruence between sound and moving pictures to discover the effect of audiovisual synchronization. The experiment used a visual stimulus of a ball on a grid surface, and a change in visual perspective of the ball coincided with auditory events. Auditory events and stimuli were based on a simple rhythmic melody played on a bass and drums (Iwamiya / Sugano / Kouda 2000: 1222-25). As a result of the experiment, when visual events synchronized with auditory events, formal congruency was perceived, and when the study used higher synchronization between events, the perceived congruence was higher. This result showed the effectiveness of formal congruency in raising perceived congruence, because the participants' rating of their impression of synchronization was also higher when the perceived congruence was higher (Iwamiya 2013).

Another example can be seen in a second experiment by Sugano and Iwamiya where they examined the effect of synchronization between auditory and visual accents (Iwamiya 2013). In this study, in addition to synchronized and asynchronized conditions, a temporal phase-shift condition between video and music was incorporated. For example, a constant time difference between auditory and visual accents occurred during the time interval between events. Their result demonstrated that the relatively higher perceptual congruence of the phase-shift condition indicated the significance of correlated periods between periodic auditory and visual events. Sugano and Iwamiya did a third similar experiment on formal congruency. In this experiment, by adjusting the speed of a visual object's movement, participants were asked to match the music presented at a constant tempo. As a result, perceived congruence was created when the periods of auditory and visual events corresponded at integer ratios, such as 1 to 1, 2 to 1, 3 to 1, or 4 to 1.

As a consequence, it was determined that higher synchronization between auditory and visual events follows a direct causation of subjects' rating an increase in formal congruency between music and video. In addition to one to one audio-visual synchronization, integer ratios of periodic patterns also cause direct congruence.

Semantic Congruency. While formal congruency is the perceived correlation of auditory and visual events, semantic congruency is the similarity between auditory and visual affects and impressions on humans. Iwamiya, Jogetsu, Sugano, and Takada examined semantic congruency using computer-generated moving objects and short pieces of music. In this experiment, participants rated the semantic differences of scenes on a computer monitor depending on congruent affect due to moving objects and music. Visual stimulus was performed under two conditions: the speed of the objects and the density of the objects. Auditory stimulus was also performed under two conditions: tempo of the music and mode of the music (major or minor). Results of this experiment revealed two types of high semantic congruency. First, high densities and fast speeds of images corresponded with major-key and fast-tempo music. Second, low densities and slow speeds of images correlated with slow-tempo and minor-key music (Iwamiya 2013).

Iwamiya and Hayashi's experiment using piano music and computer graphics of a piano performance under various colors of lighting also reveals the effect of semantic congruency. They used eight short musical passages from popular piano tunes combined with lights colored red, yellow, yellow-green, green, cyan, blue, violet, and purple. In this experiment, they examined that blue, violet, purple, and red were highly matched with minor-key and fast tempo music, while green, yellow-green, yellow and cyan were matched with bright, major-key and fast tempo music. In addition, the results of this experiment on affective impression showed that yellow, green, and cyan colors made the music seem brighter (Iwamiya 2013). Several empirical studies on formal and semantic congruency between sound and moving pictures have been discussed, providing various suggestions for producers, sound designers, and composers of visual media productions to assist in creating effective audiovisual works. However, in real applications of these suggestions to multimedia art, there are a lot more issues to consider. Furthermore, because the aforementioned psychologists' results are based on simple tasks and unified situations for drawing scientific conclusions, it could be argued that those studies cannot be directly applied to the actual creation of multimedia arts, especially that which is avant-garde. Nevertheless, if the results of the congruencies are intertwined, these studies can be qualitatively interpreted within the context of creating music and art correlating and non-correlating aspects. Details of this direct connection will be discussed more in later.

Objectives

Audio-Based Visualization in Music Performance

Audio-based visualization is a visualization driven by audio-derived data to interpret the audio in visual sense. (See Figure 1.)

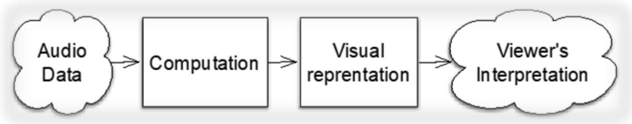

Figure 1. Audio-Based Visualization for Audio Interpretation

It is generally used for scientific work to study the audio information in visual form. The audio-based visualization of this study is a performance-oriented visualization driven by audio-based data, with the goal of helping the audience appreciate expressive music performance. (see Figure 2.)

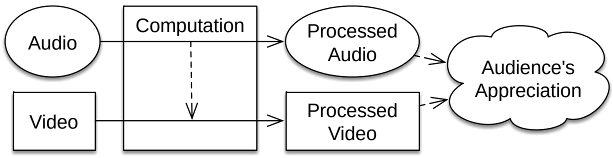

Figure 2. Audio-Based Visualization for Music Performance

Fundamental Attributes of Electroacoustic Music. Since avant-garde artists and composers push the limits of their fields, generally they are working outside of prior styles. Avant-garde artists and composers are not simply creating art to please the people's ear; they are attempting to create a style of art in a way that no one else has created before (Crane 1987: 20-25). At the same time, their new style of art does not replace the old one, instead they coexist (Benitez 1978: 53-77). Because of these reasons, the investigation of standardized characteristics of all avant-garde music would seem to be an impossible task. Even if one found a general musical rule in contemporary music, being avant-garde implies that the art form is constantly evolving and that rule may already be broken by new works. If one has only read through the table of contents of books on contemporary music, one notices that authors have conceded to the impossibility of attempting a unified approach (Benitez 1978: 53-77).

However, the definition of some general attributes of electroacoustic music deriving from the avant-garde is possible for the scope of this research. Instead of trying to unify the diverse set of musical contexts of the avant-garde, the generalized fundamental attributes can be obtained in scientific research based on physics and acoustics, research which forms the basis for much electroacoustic music. Since electroacoustic music creates musical objects from sounds that people hear in daily life, the musical sound must be defined in terms of physical phenomena which include not only pitched tones but also noisy sounds.

Amplitude and Timbre as Fundamental Attributes. In physics, sound is described as the passage of pressure fluctuations through an elastic medium as the result of a vibrational impetus imparted to that medium (Raichel 2006: 13). Because sound is a phenomenon from the nature of waves, it may be comprised of only one frequency, as in the case of a pure steady-state sine wave, or it may be comprised of many frequency components, as in the case of noise generated by construction machinery or a rocket engine (Raichel 2006: 13).

Sound may or may not be audible to the human ear, depending on its frequency content and intensity; oscillation between 20 Hz and 20kHz, with intensity (amplitude) above a sufficient decibel level, depending on the environment, will be audible. More specifically, for the sensation of sound to be audible to a human, it needs to be comprised of a certain number of aperiodic or periodic sinusoidal waves, each above a certain amplitude, which depends on the environment (Fastl / Zwicker 2007: 62-63).

A pure tone is known as a sine wave and can be described with the equation:

$$y(t) = A \sin(2\pi vt + \phi) \text{ or } y(t) = A \sin(\omega t + \phi)$$

where "A" is amplitude, "v" is frequency in Hertz, "ϕ" is the phase in radians, and the quantity $\omega = 2\pi v$ (Benson 2008: 17).

"A" is always positive by convention, and "A" can be zero. However, since "A" multiplies the sine function, if "A" is zero, there is no sound. As a result, amplitude is a primary component to produce a sound in both physical and mathematical models.

If the amplitude component is excluded from the equation, it remains a weighted-sum of a series of sinusoidal waves. Because the information of a set of sinusoidal waves determines the timbre of the sound and the pitch of the sound, that timbre and pitch are the second and the third components of sound can be inferred while amplitude is the first component.

These three essential components of sound are corroborated by the definition of sound in psychoacoustics. Poynting and Thomson categorize sound in three characteristics for human sensation: loudness "amplitude", pitch "frequency", and quality "timbre" (Poynting / Thomson 1904: 7-8). However, the definition of pitch in acoustics is differently used in music. While pitch in music is the position of a single sound in a particular tonal standard, in acoustics pitch is the relative degree of highness or lowness of a tone. In avant-garde electroacoustic music, many composers are concerned with pitch in its acoustic meaning, and in fact many composers place more importance on timbre than on musical pitch. In other words, their attention is still on the acoustic meaning of pitch, but the attention they place on musical pitch is decreased. This is a good reason to use timbre instead of musical pitch as the essential components for the universal application of the audio-based visualization.

In fact, some electroacoustic compositions are composed only with timbral aspects of sound. One may recall the concept of musique concrète, and how Pierre Schaeffer compared the aesthetics of serialism and electroacoustic music:

> Why twelve notes when electronic music has introduced so may more? Why a series of notes when a series of sonic objects is so much more interesting? Why the anachronistic use of an orchestra whose instruments are handled with such obvious anti-naturalness by Webern and his imitators? And above all, why limit the horizon of our research to the means, usages and concepts of a music after all linked to a geography and a history; certainly an admirable music but still no more than the Occidental music of the last few centuries. (Palombini 1993: 542-57)

Schaeffer pointed out the potential of sound objects as musical notes and he indicated that there might be a limitation of serial music which is still based on pitches of music. Although the evidence of the application of serial techniques can be found in Schaeffer's tape collages, one can recognize that he treated sound objects as musical notes without the aspect of pitch (Schaeffer 1950: 48-50).

In this work, and in an effort to make the accompanying software immediately compatible with many electroacoustic compositions, pitch is excluded as a general attribute of electroacoustic music that is avant-garde. Instead of pitch, timbre is chosen for the second attribute of electroacoustic music. In a manner similar to how purely pitch-based notions of dissonance have informed the syntax of tension and release in tonal music, timbral consonance and dissonance of sound objects as measured by spectral flatness can describe the tension and release in electroacoustic or timbre-based music.

In electroacoustic music, loudness is the primary component of musical sound because it has a relationship to the actual energy that makes sound, and timbre is the secondary component because it is the most elementary and convenient component that differentiates one sound object from the next. With those two components of sound, a congruency between electroacoustic music and visual processing can be formed.

Mapping Audio Data on Visuals

Previously, two components of sound–loudness and timbre–were defined and chosen as fundamental attributes of electroacoustic music. In the visualization of these components, the method of mapping is a critical matter. What visual aspect can be connected to those attributes?

What degree of mapping can best achieve congruency? For the mapping between music and visuals, a discussion of which attributes of visuals would be naturally connected to the attributes of music is demanded, and a rudimentary component of visuals should be used for the mapping.

Amplitude to Visual Brightness. Amplitude might be the easier attribute to map, as the amplitude can be applied to visual brightness in intuitive ways, so that the presence of sound corresponds to the presence of light. As the amplitude of a sound is the primary energy used to generate the sound, visual brightness can also be considered a fundamental aspect to see the visuals. Conversely, no amplitude would result in no sound, causing a lack of visuals.

Several psychological studies already reveal correlation between amplitude and visual brightness. Lawrence Marks researched the relationship enthusiastically. He wrote several articles related to the perceptual congruence between pitch, loudness, and brightness of light. Throughout his articles, he consistently argues that there is a positive correlation between them (Marks 2004; 85-102). In one of his experiments, participants in two groups, four year-old children and adults, adjusted the degree of similarity between pairs of visual stimuli and auditory stimuli. Among mixed pairs of sounds and visuals, both groups perceived the congruence between brightness and loudness and between pitch and brightness (Marks 1989: 586-602).

Noisiness to Visual Distortion. A timbre of sound was discussed as the second fundamental attribute of electroacoustic music, and the mapping of timbre to visual effect can be much more complicated. Because of an almost infinite number of possibilities of sound in real life, finding standardized mapping methods for timbre cannot easily be done. Even if the timbral information is obtained with a Fast Fourier Transform, audio visualization requires many complex mappings such as one-dimensional data to multidimensional data or multidimensional data to one-dimensional data.

Even though many other applications and studies have done audio visualization for scientific or graphical representation with spectral data, since our visualization in this study is derived from live video of a performer's movements, we don't require all the spectral information used in scientific studies. Instead, our research requires an intuitive method that can be easily correlated with visual effects for the audience's perception of congruency. In this study, Spectral Flatness Measure (SFM) of sound was employed as an evaluation of the quality of timbre. SFM is defined as the ratio of the geometric mean of the power spectrum to the arithmetic mean of the power spectrum (Johnston 1988: 318). By using SFM the value representing a sound's place on a spectrum of tonality to noisiness can be found, which can be used for measuring the timbral quality of music. In this work, for lack of a better term, the word "noisiness" is employed to describe how spectrally flat a signal is. A signal with the highest measure of noisiness (1) is white noise, and a signal with the lowest measure of noisiness (0) is a pure tone.

There are many ways that the visual image can be altered according to the timbre of sound for audio-based visualization. For simplicity, in this study, it has been decided to use the mass spectral flatness to blur or distort the visuals. Since the value of SFM is representing the degree to which a sound is tone-like as opposed to noise-like, it can be correlated with the noisiness of visuals. In other words, the quantity of the noisiness in the sound can be associated with the level of distortion of the visualization. Subsequently, the value of noisiness by SFM is connected to the value of the distortion of the visualization. Depending on the type of visual effects, the final result of the visualization can be varied. Detail of the visual effects by noisiness will be discussed in Implementation section.

Visualization of Expressive Body Movement

Because the number of video cameras determines a variety of visuals, the proper number of cameras should be contemplated, and because the placement of the camera is directly related to the visuals' perspective of the performer, each camera must be positioned with careful consideration for the final result of the visualization. In addition to camera arrangement, lighting is an essential element in the visual presentation. By projecting appropriate lighting, clear contrast of the light and shade can be created, and by clear contrast, several levels of silhouette can be produced. These three components—the number of cameras, the placement of cameras, and the manner of lighting—must be taken into consideration.

Multiple Video Camera and Placement. Multiple Perspective Interactive Video (MPI-video) is a strategic system that allows a viewer to view an event from multiple perspectives interactively, even based on the contents of the events (Katkere 1995: 202). The idea of MPI-video has been useful for various research such as immersive video for telepresence systems, traffic monitoring and control, and the analysis of physical performances, sports and dance (Katkere 1995). Using MPI-video, a set of sequences is obtained from an object or event, and used to build a multi-dimensional model of the important events in a scene. In the case of audio-based visualization, the information of audio decides the best camera angle at every time instant, while other systems consider user request for camera selection (Katkere 1995).

One of the purposes of this research is to project the performer's expressive body movement in multiple perspectives which cannot be easily viewed from the audience. Depending on the placement and angle of the camera, the expressiveness can be magnified or reduced. If the placement and angle is not in an appropriate position, the cameras might not contribute to the visualization of the performance. Thus, the composer should carefully place and angle the cameras, according to the final visualization on the screen.

There are two aspects to be maintained when the cameras are positioned for the performance. The first aspect is finding suitable locations to place the cameras. A location is acceptable if it meets the following three requirements: the camera is not blocking the view of the audience, the camera view is not blocked by the equipment or instrument, and the camera is not disrupting the performer's movement. If the camera is placed in a suitable distance from the performer, and if the space between performer and audience is not disrupted on the stage, a feasible location might be found.

The second aspect is the angle of the cameras. The angle of the cameras should be varied depending on the final aim of the visual effects. If multiple camera views are to be projected in a distinct place on the screen, the orientation of the subject in the views should be separated by physically adjusting the angle of the cameras. For example, this would be necessary if the performer's hand is positioned on the left side of the screen, and the performer's face is placed on the right side of the screen. However, the positioning of each view can be controlled computationally, which is not considered in this study, but even in this case physical positioning on the proper area of the screen is needed.

Figure 3. Comparison of the overlapped view and juxtaposed view. Left: Two views are overlapped in the center, Right: Two views are juxtaposed.

Since the brightness of each camera view is controlled by the amplitude of audio processing, each subject of the view can be separated depending on angle of cameras and the loudness of each audio effect. However, because there is a possibility that multiple views can be overlapped simultaneously, depending on the composer's intention the separation of views of subjects might be necessary. (See Figure 3.)

In addition, the final decision of equipment setting was based on two criteria; whether the final visuals displayed each gesture and its connection to sound, and whether each video was adequately magnified to show the expressiveness of gesture. For example, when each view of the gesture overlapped and it obscured its connection to sound, the positions of each view were juxtaposed. When expressive body movement is too small to be recognized, it is magnified and overlapped in the center.

Three Point Lighting and its Usage. In video production, quality lighting is one of the most important elements. When everything is brightly lit with a fine use of lighting and shadow, the best images can be made (McIntosh 2012: 3). In the actual application of lighting for images, several factors were considered: ambient light, white balance, softness or hardness of light, light bouncing, shadow, and depth of subjects. Among them a couple of factors can be modified physically by the placement of lighting. More specifically, "three-point-lighting," the standard ideal lighting arrangement for video production was adapted. In this system three lights are required: a key light that is the hardest and brightest light on the subject, a fill light that is a soft light to fill the shadow cast by the key light, and a back light that visually differentiates the subject from the background. By using three separate positions, the subject can be illuminated properly, and the desired fine images or vivid videos can be achieved.

In the performances of the compositions, the usage of a fill light was not implemented. Because the fill light faces toward the audience, it might distract the view of the audience. Another reason is that the fill light might eliminate the shadow which creates "depth" and "emotion" in the images (McIntosh 2012: 12). Only the key light and the back light were used.

While the key light can be placed in front of the performer, the back light can be created by using the house lights of the concert hall. Most concert halls have their own house lighting system to point at the performers, one can take advantage of the house system. The combination of the house light and point light can help to illuminate expressiveness of the performer's motions. Depending on the direction, house lighting can be used as key light or back light. During the preparation of the performance, different lightings using the adjustable portable light were attempted and compared, and appropriate lighting arrangements were chosen by the composer. More details on the lighting for each composition is discussed in Implementation section.

Formal and Semantic Congruency. Previously, two kinds of congruency in the psychological research were reviewed: formal congruency and semantic congruency, and the amplitude and noisiness applying to visual brightness and distortion were discussed. By connecting the data of the amplitude to visual brightness, the formal congruency between the visuals of performer's instrumental gesture and instrumental sound can be created. By connecting the noisiness to visual distortion, the audience can feel another Semantic congruency between timbral dissonance and visual distortion.

However, some portion of expressive body movement is missing in the visualization. Since in the audio-based visualization the audio data is controlling the brightness of visuals, audio always precedes visuals. It means that some portion of expressive body movement cannot be seen through the visualization when there is no background sound or no sustained instrumental sounds previously. In other words, the audience cannot see the expressiveness of the performer in the visualization if there is no sound. Nonetheless, this missing portion of expressive body movement is not considered as important by the author, because in the actual performance it was too subtle to affect the congruency of the particular performances for this work, and it can be seen only in the beginning of the performance.

Implementation

Audio and Video Processing

Designing the audiovisual system requires a specified methodology to implement the objectives. Even though a concrete theory is well-defined, without the example of a real application many details and issues might not be discovered. Cognitive scientists Minsky and Papert indicated the importance of real cases in their book:

> Good theories rarely develop outside the context of a background of well-understood real problems and special cases. Without such a foundation, one gets either the vacuous generality of a theory with more definitions than theorems—or a mathematically elegant theory with no application to reality. Accordingly, our best course would seem to be to strive for a very thorough understanding of well-chosen particular situations in which these concepts are involved. (Minsky / Papert 1988: 3)

They indicated the importance of the real application and actual implementation. The one of the purposes of this paper is the specific implementation of audio-based visualization and its performance. From the general scheme to the actual usage, the specification and methodology of audio-based visualization is discussed, and, from software to hardware, the issues and problems of audiovisual processing are examined.

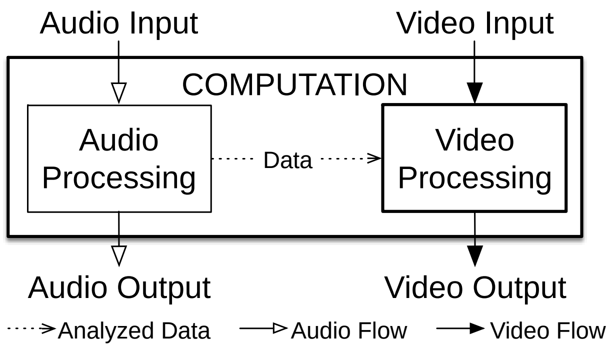

Figure 4. Overall Scheme of Audio-Based Visualization for Live Music Performance

Three electroacoustic compositions, *Piece for Solo Snare Drum and Electronics, Memoriam,* and *Synesthetic Moment,* are examined as examples of audio-based visualization. The audio processing which controls video processing is discussed, and the concept of the programming in Max 6 (Zicarelli 1998: 466) is examined. Figure 4 shows the overall scheme of the programming of audio-based visualization.

Audio Processing Units. The Max 6 audio programming environment was used for the digital signal processing of the three compositions.

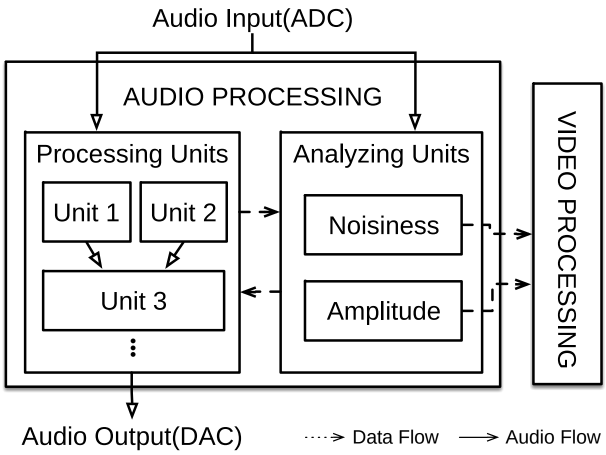

Figure 5. Overall Scheme of Audio Processing: Signal Flow of Processing Unit and Analyzing Unit.

Inside of the programming there are two distinct units: the processing unit and the analyzing unit. The processing unit consists of several sub-units including recording, playback, delay, pitch-shift, granular synthesis and spatialization. The analyzing unit is comprised of two sub-units, a noisiness unit and an amplitude unit. Figure 5 illustrates the overall scheme of the audio processing.

In the processing units, there are a couple of Max bpatchers (imported patch encapsulation) designed by the composer. The first bpatcher, “10tapsPitchShift” created for a customized delay effect and pitch-shift of which parameters can be manipulated differently. This bpatcher can produce up to ten distinct delay sounds, and all the parameters, such as delay-time, feedback-rate, and interval of pitch-shift, are randomized within a certain range of numbers. When the amplitude of the input sound passes a certain level, the analyzing unit sets the delay-time of the sound and the interval of the pitch-shifting and triggers to play it. The delay units of the bpatcher do not loop constantly, but play back only certain times in short delays (0-500ms) and irregular feedback rates. Sounds are then sent through the “gizmo~” object, which performs frequency domain pitch shifting on the sound. Even though a standard delay effect technique is the main signal processing of this bpatcher, the actual output does not sound like a delay effect because all the parameters are manipulated differently by randomized numbers. This bpatcher was commonly used throughout three compositions.

The second bpatcher, “record_reverse”, was made for the function of recording a sound into a buffer and playing it back at different rates with or without pitch-shifting. The recording is activated by the amplitude of live sound input, and playback is triggered either at the end of the recording or by the events of the compositions. This bpatcher consists of “record~”, “buffer~”, “groove~” and “gizmo~” objects. The “record~” object records the input sound into “buffer~” object, and the “groove~” objects play the recorded sound by playback rate. “gizmo~” shifts the pitch by FFT data. This patch is only used in the third piece, Synesthetic Moment.

A third bpatcher, “4chTimbreSpatial”, is also only used in the third piece. The main function of this bpatcher is to diffuse sound through a multi-channel speaker array, and the current version of the patch includes a quadraphonic speaker arrangement. In the bpatcher, a couple of objects were used for spatialization, for example the “cross~” object separates high and low frequency sounds at the crossover frequency or cut-off frequency, and the two separated sounds are spatialized in different positions. Two separated sounds were controlled by randomized position with spherical coordinates: azimuth, elevation, and distance. Two objects made at the Institute of Computer Music and Sound Technology, ambiencoder~ and ambidecoder~, were used for the spatialization. The function of these objects is to allow encoding and decoding using real-time spatial data of virtual sound sources and speaker configuration in three dimensions of up to third order B-format ambisonics (Schacher / Kocher 2006).

Besides those bpatchers, several audio processing patches were employed, including a delay effect with pitch-shifting, spectral synthesis, reverberation, and granular synthesis with the idea of a sound particle cloud. For delay effect with pitch-shifting, “+pitchdelay~” made by Tom Erbe was used for all the compositions, and the sound of this object can be heard in several parts of the music. The function of the object is to delay a sound with continuous pitch-shifting. More specifically, each delayed sound has a different pitch as time moves forward, either consistently increasing or decreasing. For spectral synthesis, “tap.spectra~” object from the tap.tools object collection which is created by Electrotap was employed. The object remaps the spectral data from FFT bins to different bins of an IFFT. For instance, if 100hz - 500hz as source range and 500hz - 900hz as destination range, frequencies from 100hz to 500hz of the sound are shifted up to the destination range of 500hz to 900hz. As an output, depending on the parameter of frequency range, pitch and timbre of an input sound can be modified to create different sounds.

Granular synthesis of a sound particle cloud was used in the first composition. Using the randomized particle within the range, the stochastic delayed sound is produced (Roads 2001), and the delayed sound represents the virtual performer in the piece For granular synthesis, “gran.cloud.live~” object created by Nathan Wolek was employed. One of the purposes of the composition is to produce a virtual performer’s sound, and usage of this object in the piece is to sample the actual snare drum sound and reproduce the sound as granular synthesis. Because a percussive sound has a fast attack, an exponential decay envelope was used for windowing each grain.

Reverberation with different parameters was used in three compositions. Two reverberation objects were used: “newverb~” made by CNMAT of University of California at Berkeley, and “yafr2~” designed by Randy Jones from Madrona Labs. Besides the objects, a reverberation with over 30 seconds was demanded for the musical concept of the second piece. The “yafr2~” object was customized to achieve a longer reverberation time by changing its internal feedback parameters.

Audio Analyzing Units. There are two components derived in the audio-analyzing unit—noisiness and amplitude—which provide two functions for influencing the processing units. The first is to send data to the video-processing unit for manipulating the visual effects dynamically, and the second is to trigger audio effect events in the audio-processing unit. The triggering can be decided by the amplitude of sound or the velocity of sound impact. For analyzing the noisiness of a sound, the "noisiness~" object created by Tristan Jehan was used, and a built-in object "peakamp~" was used for amplitude data.

There are a couple of issues in analyzing audio data. Because several "noisiness~" and "peakamp~" objects were implemented along with various audio effects, the computer slowed down, or had high latency in the processing. This issue can be solved by changing the polling interval of the objects. After trying many different intervals, a number above 50 milliseconds was quite stable for the visualization. There is another reason to choose 50 ms as the frame rate, which will be discussed in the next section.

For better performance of visual processing and audio processing, it is necessary to minimize computational delay or latency. In the "noisiness~" object, one of the arguments is the order of onset, and by providing different arguments each object can be separately executed in succession, which reduces the possibility of slowing-down the CPU. Because there are several ways to analyze amplitude, several objects were tested to find out which is most efficient and least computational. Max objects "levelmeter~", "meter~", "live.gain~", "avg~", "average~", and "peakamp~" were examined, and "peakamp~" was less computational than the others by up to 20% when 1000 copies of the objects were executed.

Magnification and Augmentation of Live Video

The visualization requires specific circumstances and particular programming which are related to audiovisual mapping, visual processing, and installation of visual equipment. Through experimentation, various methods were examined. According to the varying successfulness of those methods, the most efficient method was determined.

Mapping Audio Parameters to Visualization. The visual-processing unit receives data on the amplitude and noisiness of sound from the audio-analyzing unit. After receiving the data, the visual-processing unit converts amplitude and noisiness values to visual brightness and distortion values. Since each sensation has different perceivable measurements, the transformation of the data to the correct scale is essential; logarithmic decibels of sound should be changed to linear numbers of visual brightness, and the spectral flatness value should be transformed to the percentage of visual distortion, which can be represented by the number from zero to one.

For the mapping of amplitude to brightness, two steps of unit changing and one of scaling were applied. The first step is to map the exponential amplitude data of "peakamp~" to the logarithmic scale of human hearing, which more closely approximates the audiences' perception. The second step is mapping the logarithmic decibel data to the linear value of brightness. The third step is the scaling of the minimum value of amplitude mapping, which is optional depending on the overall situation of brightness. For instance, when a motivic sound is played with loud background sounds, the visualization of the motivic sound is hardly recognizable, because the loud background sounds raise the brightness of all the visuals. This issue results in no congruency of motivic sounds. In this case, additional scaling of the loud background sound is required. The function of the scaling is to align the brightness and perceived amplitude by changing the minimum value of the units.

Audiovisual Synchronicity. According to Van Eijk, Kohlrausch, Juola, and Van De Par, cross-modal events will be perceived as simultaneous, when the audio component precedes the video by fewer than 50 ms or the audio follows the video by less than 150 ms. The range is called point of subjective simultaneity (PSS) by the authors. More research on the perception of synchrony is Lewkowicz's article (van Eijk, Kohlrausch, Juola & van de Par 2009: 1254-63). In this experiment, he compared adults and two to eight-month old infants, and discovered the threshold for perceiving asynchrony. When the sound preceded the visual, adult participants perceived asynchrony starting at 65 ms, and when the visual preceded the sound the threshold was 112 ms, while the infants required less than 350 ms and 450 ms of difference, respectively. Table 1 show the audio range of perceived synchrony from the research.

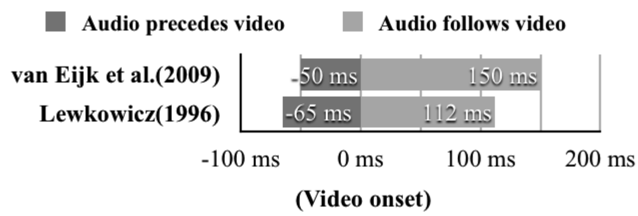

Table 1. Range of Perception of Synchrony by Audio onset or PSS.

Because part of this research is to examine the way to manipulate visuals with sound, audio processing is executed first, and the visualization is operated by the data from audio-analyzing unit afterward. Thus, the numbers after the visual onset are not related to this study. (See Table 1.) 50 ms was selected to be the interval to capture audio information, because it is the maximum acceptable number and meets the conditions of the perceived synchrony of the two researchers when there is no computational delay.

Visual Effects and Processing. For computational effectiveness, the visualization is mostly programmed with

OpenGL (Open Graphics Library). OpenGL, developed and optimized by Silicon Graphics, Inc., is a software interface for graphics hardware (Wright / Sweet 1996: 15). More specifically it is an interface that your application can use to access and control the graphics subsystem of the device upon which it runs (Seller, Wright / Heamel 2014: 3). A modern day computer assigns complex visual processing to its Graphics Processing Unit (GPU) to produce images for the display with its own graphics memory that is called Video RAM or VRAM. Current GPUs consist of a large number of small programmable processors called *shader cores* which run mini-programs called *shaders*, and from a few tens to a few thousand of these cores contained in the GPU greatly increases the processing of graphics by parallel computing (Seller / Wright / Heamel 2014: 5). Because OpenGL is mostly executed by the GPU, by using GL objects in Max 6, video-processing of the compositions can be carried out mostly on the GPU. As a result, the visualization is executed effectively, and does not interrupt the audio-processing.

The usage of GL objects in the compositions were programmed in 2-dimensional textures with OpenGL. The textures are rendered on planes facing the screen with z coordinates set to zero. In OpenGL graphics, there are two stages that are programmable: vertex processing and fragment processing. Since this project is dealing with the video from the camera as 2-dimensional textures, only the texture was modified by loudness, and noisiness in fragment processing.

In Max 6, OpenGL graphics are accessed by GL objects, which are named with the prefix “jit.gl.” A streamlined interface of the objects performs the general purpose of GPU-based grid evaluation, with each *shader* object processing the 2D texture mapped onto a 3D object. In this way, a 2D composite image can be created highly efficiently. By using the “jit.gl.slab” object, the customized texture processing in 2-dimensional view can be executed, and by referencing *GL Shader Language*(GLSL) files as an argument, the customized visual effect based on GLSL programming can be displayed. In the three compositions, three built-in *shaders* and one customized *shader* were implemented. Figure 6 shows the arrangement of the *shader* and data flows.

For the connection between amplitude and visual brightness, a function to control the luminance of each video source is required. To accomplish this, a customized shader was employed, which was programmed by the composer. When using the shader, the value of the amplitude data is multiplied with the value of luminance, and consequently the brightness of the video is manipulated by the amplitude. The “jit.gl.slab” object referencing the customized shader file can handle up to eight inputs of videos, and in this project, three video inputs were implemented.

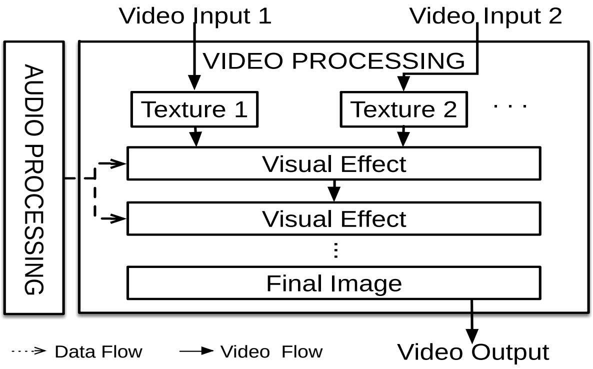

Figure 6. Overall Scheme of Video Processing.

The second goal is to connect the noisiness to visual distortion in an artistic way. The artistic goal with this type of processing is to find a visual boundary between clarity and distortion of the video. When the noisiness is high, visual distortion is maximized, and when the noisiness is low, a clear view of the video is displayed. However, if the visual is excessively distorted the expressive body movement cannot be seen, but without enough distortion no congruency is perceived. Thus, the type of visual effect and the degree of distortion are critical. Among many other built-in shaders, three shader effects were implemented.

The first shader is “td.lumadisplace.jxs”, which provides a visual effect that spatially displaces the texture by the luminance. There are two parameters to control the effect: offset, and amplitude. Each parameter needs two float numbers (-1. to 1.) to determine the X and Y coordinate value for the direction and the width of the displacement from the center of picture. Figure 7. shows the difference between a processed image and unprocessed image.

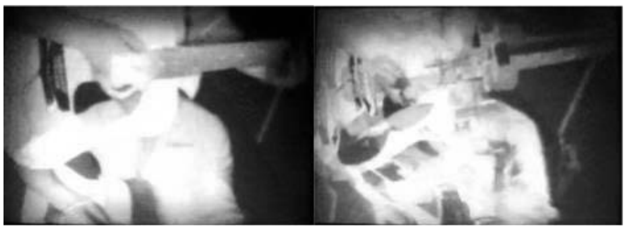

Figure 7. Views with or without Luma Displacement Processing. Left picture: image without luma displacement processing, Right Picture: image with luma displacement processing controlled by noisiness value.

The second shader is “tr.edgeblend.jxs”, which generates gradient alpha for edge-blending. This object requires a parameter that consists of four float numbers. Each float number represents the width of edge-blending of each edge in a 2-dimensional view. The function of this object in the visualization is not only to focus the audience's vision on the main subject in the video, but also to eliminate the opaque rectangular edge of the video, which might look less artistic.

The third shader is "td.sinefold.jxs", which results in a ripple-like spatial displacement. By controlling several parameters such as center position, frequency, amplitude, and phase of the ripple shape, a wavy-surface texture of the image can be created. This object was used for visualizing the spatialization of sound in Synesthetic Moment, the third composition. The coordinate of the sound source from the spatialization bpatcher was sent to control the center position of the wave effect. The values of frequency, amplitude and phase of the ripple effect were fixed. As a result, the randomized movement of sound from the bpatcher is presented on the screen by the movements of a wavy texture. The difference can be seen in Figure 8.

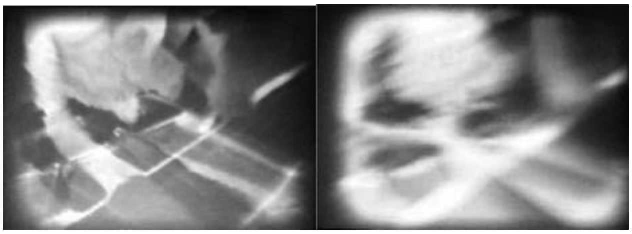

Figure 8. Views with or without Sinefold Processing. Left picture: image without sinefold processing, Right picture: image with sinefold processing controlled by the spatialization of sound; position of the sound source is middle-left.

Installation of Video Cameras and Lighting. Depending on the lighting situation of the concert hall, the installation of video cameras and lighting is varied. The cameras and lights of each piece were differently placed in the performance according to three-point-light technique.

For the first piece, the cameras were placed in three designated spots: (1) the right side of the performer, (2) the left side of the performer, and (3) the left side of the performer to point from drum to torso. The first two cameras capture the close-images of hand movements from approximately a 45-degree angle in front of the performer. The third camera aims at the upper body of the performer from the side. Figure 9. shows the placement of the cameras in Piece for Solo Snare Drum and Electronics.

The spotlight of the first piece was placed in behind and to the right of the performer. In this case, the spotlight is used as back light, and the house light is employed as the key light and fill light. The lighting of the first piece is illustrated on the right side of Figure 9.

In the second piece, Memoriam, three cameras were employed, each in one of three different angles. The first camera angled from the instrument to the legs of the performer. In one version of this piece, the performer crushes the instrument at the end of the piece, completely destroying it, which is not notated in the score. The second and third cameras pointed at the upper body and hands of each performer. (see Figure 10, left)

The lighting of the second piece has a similar setting as the first piece, but the spotlight was used as a key light to illuminate the expressive body movement and the instrument. The house light was used as back light. (See Figure 10. Right.)

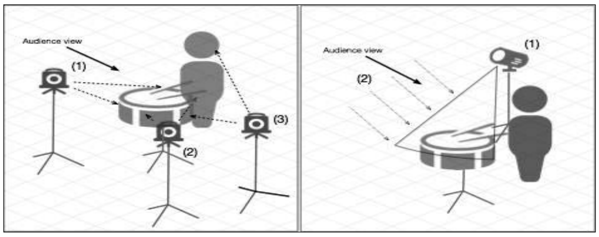

Figure 9. Placement of multiple cameras and lighting for the first piece. Left: Placement and angle of multiple cameras; (1) Point hands from right side, (2) Point hands from left side, (3) Angle from drum to torso. Right: Lighting; (1) Spotlight from the back, (2) house light from audience side.

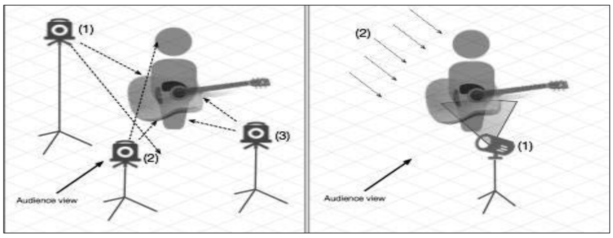

Figure 10. Placement of multiple cameras and lighting for the second piece. Left: Placement and angle of multiple cameras; (1) Angle from guitar to legs, (2) Angle from auitar to face, (3) Angle to hands. Right: Lighting; (1) Spotlight illuminates the performer as key light, (2) House light brighten the whole body of the performer from different angle.

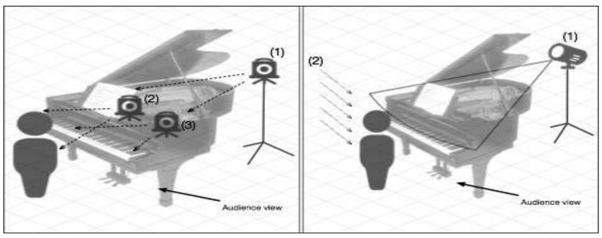

Figure 11. Placement of multiple cameras and lighting for the third piece. Left: Placement and angle of multiple cameras; (1) Angle to soundboard, (2) Angle from upper body to face, (3) Angle to keyboard. Right: Lighting; (1) Spotlight illuminates the performer and piano as key light, (2) House light brightens the whole body of the performer from the back.

In the third piece, Synesthetic Moment, the placement of the cameras was limited. Because of the open space for the audience's view and the structure of a piano, the cameras were located in similar directions from piano to performer. The first camera pointed at the soundboard of the piano for capturing the hands when the performer plays extended techniques on strings. The second and third were placed on the music rack of the piano. The second camera was angled towards the upper body and face of the performer, and the third one was tilted towards the keyboard for capturing hand gestures. (See Figure 11, left.)

Because the top board of the piano blocks the ambient lighting, the spotlight as key light is significant. The key light illuminates the body and face of the performer and the piano as well. The house light as back light decorates the hands and the edge of the performer. (See Figure 11, right.)

In each composition, key light illuminated the expressive body movements in the way of three-point-lighting, and house light was used as back light. Three cameras were placed in each composition depending on the instrumentation and performer's position. By applying proper settings of lights and cameras, the visualization of expressive body movements was maximized, and the distraction by equipment on stage was minimized.

Evaluation and Conclusion

Composer's Evaluation

From several performances in different venues prior to the doctoral recital, a number of strong and weak aspects were discovered. By addressing the problematic aspects, the system of the compositions was gradually improved. The most recent versions of the three compositions were performed in the doctoral recital. Based on the experience from the performances, various aesthetic values and technological values were obtained.

Aesthetic Value. Four advantages of the audio-based visualization are presented which contribute to the aesthetic value of composing with the audio-based visualization. First, the composer does not need to lose the autonomy of the musical content while still augmenting the performance. In other words, the musical content is not changed by the visual component in this method. Because electroacoustic composers are concerned about sound and musical content as the core of the work, it is important that any other component not interfere with the sound of the music itself.

Second, the author believes that the visual property of expressive body movement contributes to convey the performer's intention to the audience, making it more expressive and understandable. By magnifying performance movements, the performer's method of communicating intention can be augmented, hopefully enabling the audience to better understand the musical intention. For example, at the moment the performer is articulating an important phrase with his or her wrists, an audience may see the zoomed-in detail of expressive wrist movements. It has the potential to result in a more understood performance, even if the expressive body movement is small.

Third, the audience can perceive musical aesthetics in the visualization. Since the syntax of tension and release in electroacoustic music has been converted to a visual representation, the audience can potentially appreciate musical aesthetics in two sensations simultaneously. For example, the increasing dynamics in a phrase creates a musical tension, and the dynamics are perceived by increasing the brightness of visuals. As a result, the visual representation of musical contents creates an incorporated sensation.

Fourth, audio-based visualization is adaptable for other compositions. Because the general characteristics of sound are applied in the visualization, the system can be implemented in any composition, and each performance results in different visualization. For example, the source of visuals will be different by providing different stage settings and instrumentation, and the visual effects will be diverse depending on musical content and audio data.

Technical Notes. In the initial trial of the system, two laptop computers were employed: one for audio, and the other for visuals. However, because of the complexity of both hardware and software settings, the system was modified to run on a single laptop computer. Finding effective methods of programming was paramount, and three technical values related to the method were found.

First, an effective method of time synchronization between audio and visuals was achieved. According to the point of subjective simultaneity in psychology, the delay time in audio analysis and video processing were tested, and ideal numbers were determined.

Second, an algorithmic method to map musical attributes to visual attributes was introduced. Because music and visuals are two separate sensations, the mapping requires several scalings depending on musical contents. For instance, in the case when there is a loud drone-like sound in the background with a relatively soft theme, which would otherwise result in the visual brightness of the drone sound overwhelming the visualization, the brightness needs to be additionally scaled in a low range. The same case would apply to a loud theme, in which case the brightness would be scaled to the middle range.

Third, a method to improve visual processing was examined. CPU-based visual processing was converted to GL objects, and unnecessary visual effects were eliminated or replaced. Moreover, the scheme of visual processing was arranged and simplified.

Procedure and Results of Survey

Procedure. Surveys were processed during the author's recital, and the recital was open to the public. Three electroacoustic compositions were performed without an explanation of the objectives of this research, and then a lecture followed the performance. The lecture was given following the performances in order to help avoid biasing the perception of the participants according to the opinion of the author. After each performance, the audience filled out a questionnaire about the composition.

Forty-two people attended the concert, and everyone participated in the survey. Among them, twenty-five people were male and seventeen were female. The average age was twenty-nine years with a standard deviation of 13 years and 6 months. Participants were sorted into two groups by categorization of their musical background: years of musical training, the number of avant-garde music concerts they had previously attended, and the number of electronic music concerts attended. The participants who have been trained more than four years were twenty-five, while seventeen people had musical training of less than four years or none. Twenty participants had attended an experimental music concert once or never, and the participants who had attended more than one numbered twenty-two. Half of participants had no experience, or had attended electronic music concerts once, and the other half had attended two or more.

Hypotheses. There are four hypotheses in the survey. The first hypothesis is that the audience can perceive congruency between amplitude and visual brightness, and between noisiness and visual distortion in the audio-based visualization in the three compositions. Second, the audio-based visualization is helpful to appreciate the music performance and expressive body movement. If the first two hypotheses are true, these two hypotheses become the assumption of the third and fourth hypotheses. The third hypothesis is that the degree of perception for the congruencies of loudness and noisiness is different depending on musical background, and the fourth is that the degree of appreciating the performance and expressive body movement is different depending on musical background.

Based on these four hypotheses, a questionnaire was made with the following factors: (a) musical background of the participants, (b) perception of the congruency between visuals and music, and (c) whether the visualization is helpful for music appreciation and expressive body movement. By these three categorizations of the musical background of the participants, the degree of perceived congruency and appreciation of the performance can be compared within the groups.

Results of Survey. For the first and second hypotheses, we are making an assumption that the audience feels neutral and experiences no congruency on each category of the hypothesis in traditional music performance. Each question in the survey is based on a premise of the hypothesis, and a one sample t-test for each question was applied. Collectively, the answer of each question is statistically significant, ($p<0.01$) and the result can be interpreted as the audience perceiving a congruency. As a result, the null hypothesis, which states there is no significant congruency, is rejected, and as the alternative hypotheses, the first and second hypotheses are accepted: the audience can perceive the congruency between sound and visuals in the audio-based visualization in all three compositions, and the audio-based visualization is helpful to appreciate music performance and expressive body movement, assuming that the audience would provide neutral ratings for each questionnaire question in the absence of the visualizations.

For the third and fourth hypotheses, the number of participants is questionable and the population is not clearly Gaussian. For better results the Mann-Whitney U test, which is one of the nonparametric tests, was applied, in which the assumption of normality does not need to be satisfied. Depending on the p-value of a Mann-Whitney U test, whether the third and fourth hypotheses are true can be proved.

For the evaluation we need to separate the participants into two groups depending on musical background, so three factors of musical background were applied: length of musical training years, experience of avant-garde music concerts, and experience of electronic music concerts. There are 17 participants who have musical training of less than 4 years, and 25 people have more than 4 years of musical training, so 4 years is chosen as our threshold to describe musical experience. Additionally, 20 participants have attended an avant-garde concert less than 3 times, and 22 have experienced an avant-garde concert 3 times or more. For the electronic music experience, half of the population has attended the concert 3 times or more, and another half, less than 3 times, so 3 years is chosen as our threshold for experience with avant-garde music.

The third hypothesis is that there are differences between the groups for the perception of the congruency. There were three questionnaires for congruency in the survey, between visuals and loudness, visuals and noisiness, and visuals and body movement. The mean value of most of the answers is around 5 or more degrees (1 = disturbing, 7 = high congruency), but only one response is statistically significant. In the grouping of the experience of avant-garde music concerts, there is a distinction in the congruency between loudness and visuals: group one (less familiar with avant-garde music) has M=5.52, SD =1.2, and group two has M=5.98, SD=1.02, resulting in a p-value of 0.012. As a result, the null hypothesis is rejected, that there is no significance between those who have seen several avant-garde concerts and those who have not. The result can be interpreted as people who are more familiar with avant-garde music perceived higher congruency between loudness and visual brightness. (See Table 2.)

		Musical Training years			Familiar with Avant-Garde music Concert			Familiar with Electronic Music Concert		
		< 4 years	>= 4 years	p-value	< 3 times	>= 3 times	p-value	< 3 times	>= 3 times	p-value
Loudness & Visuals	Mean (SD)	5.67 (0.93)	5.81 (1.27)	0.339	5.52 (1.2)	5.98 (1.02)	*0.012	5.74 (1.13)	5.83 (1.15)	0.307
Noisiness & Visuals	Mean (SD)	5.22 (1.22)	5.14 (1.55)	0.415	5.2 (1.41)	5.15 (1.42)	0.452	5.27 (1.32)	4.74 (1.74)	0.167
Body Movement & Visuals	Mean (SD)	5.04 (1.51)	4.79 (1.67)	0.189	4.92 (1.78)	4.82 (1.43)	0.416	4.97 (1.65)	4.43 (1.41)	0.267
*p<0.05										

Table 2. Means and Standard Deviations for the congruency between sound and visuals in three different groupings.

Since the rest of the questionnaire has a p-value greater than 0.05 ($p > 0.05$), the responses have a high chance that random sampling would result in the mean randomless being far apart as observed in this experiment. In these cases, more evidence or samples would be demanded to find significance. The fourth hypothesis is that there is a distinction between halves of our three groups (those with and without musical training, avant-garde experience, and electronic music experience) for appreciating the performance and feeling the expressive body movement in the three compositions. Three discriminations were applied, and two questions were asked. Two questions are statistically significant in the grouping of length of musical training. The first question regarded whether the audio-based visualization helped the participant appreciate the music. The group with less than 4 years of musical training has M=5.47, SD=1.76, the other group with more than 4 years has M=5.14, SD=1.38, yielding a p-value of 0.021. The second question is whether the visualization amplify the expressive body movements. The group for less than 4 years has M=5.71, SD=1.22, and the other group with more than 4 years has M=5.13, SD=1.50, yielding a p-value of 0.003. This result can be understood as people who have less musical training years feel more helped by audio-based visualization to appreciate the performance and amplify the expressive body movement. In other words the duration of musical training can affect the degree of the music appreciation of the performance of three compositions. (See Table 3.)

		Length of Musical Training			Familiar with Avant-Garde music Concert			Familiar with Electronic Music Concert		
		< 4 years	>= 4 years	p-value	< 3 times	>= 3 times	p-value	< 3 times	>= 3 times	p-value
Help Appreciate	Mean (SD)	5.47 (1.76)	5.14 (1.38)	*0.021	5.15 (1.76)	5.42 (1.3)	0.333	5.3 (1.57)	5.22 (1.48)	0.378
Amplify Body Movement	Mean (SD)	5.71 (1.22)	5.13 (1.50)	*0.003	5.4 (1.46)	5.35 (1.37)	0.348	5.53 (1.39)	4.74 (1.32)	0.195
*p<0.05										

Table 3. Means and Standard Deviations of Appreciation of the performance or ampligying the expressive body movement.

In summary, although there are several statistical matters such as populations and normal distribution that could be improved, a couple of valuable results are significant for audio-based visualization and its application. First of all, audio-based visualization might be useful to help the audience appreciate the music performance, and the method of the audio-based visualization can convey performer's expressiveness between visuals and music. Second, while people who have more experience of avant-garde music perceive congruency between loudness and visuals, these people who have more musical training feel that this method is less helpful to appreciate music performance than those who have less musical training. It is worth noting that the subject test participants were attending the author's thesis defense and may had a non-neutral bias in favor of the author's work. In future work, it would be nice to verify the work by specifically recruiting impartial participants and to carry out control experiments to verify how subjects would answer the questions in the absence of any digital visualizations. Nonetheless, the present survey results appear to positively support the concept of audio-based visualization. In particular, its advantages seem to include the following:

1. the method of the visualization is helpful to appreciate the music performance,
2. the audience can perceive the expressiveness of performer's body movements through the visualization,
3. the loudness and noisiness of audio can be a persuasive option to visualize music performance.

Future Work

Developing and evaluating audio-based visualization has provoked new compositional ideas and technical plans for future research, deriving from issues and values relevant to the research. These ideas are outlined in the following sections.

Compositional Idea. Although the benefits of audio-based visualization have been demonstrated in the form of three compositions, the idea could be expanded in several ways. First of all, prerecorded video content could be incorporated. By adding pre-recorded videos, the audience could enjoy and appreciate more variety in the visualizations, and at the same time composition could have further options for expressing metaphors in the composition via the pre-recorded video. It contributes to broaden the enormous possibility of audio-based visualization, and to offer a component of programme music, which invites imaginative correlation with the music.

Second, a composition for theme and variations could give a demonstration of diverse visualizations which gradually change. By developing the musical motif in each variation, the visualization may be gradually changed analogously, and/or by varying visual effects, the different motivic ideas of each variation could be conveyed to the audience. For example, in a rhythmic variation, the time synchronization between audio and sound could be increased by reducing the delay time of processing, and in a harmonic variation, the spectral information of sound could influence the visual effects. As a result, various musical content of each variation would convert to diverse visual representations.

Technical Setup. Different types of cameras can enlarge the possibility of the visualization. For example, a motorized camera can capture different angles of video depending on the audio or gestural data. The motorized cameras may move around in a designated spot and finds the proper spot for a better view by the data. A high-dynamic-range camera also can be implemented for better images with more details even in the situation of insufficient lighting. A thermographic camera, also called an infrared camera, is another option to experiment in the situation without lighting.

Another plan is to increase the number of lights. In the performance of the three compositions, only one spotlight was employed as key light to illuminate the performer, and house lighting brightened the rest of the subject including instrument and background. However, if other spotlights are added instead of house lighting, the visualization can be more adjustable and satisfying.

The positioning and magnification of the visualization can be dynamically changed by mapping of the values. For example, when a sound is shifting to the left channel, the position of the visualization is moving toward the left side of the screen. When the amplitude is decreasing, the visualization is gradually magnifying the gestures to display the small gestures. As a result, additional dimensions of visuals could be amplified for different visual representations.

Finally, various psychological studies of multimedia art are possible. A study of musical noisiness and its perception is rarely found, and a study of realtime responses of audience's perception that can be realized by using a web-based interface such as NEXUS are important for future research. These studies can broaden the possibility of multimedia art, and the potential of visual-music performance.

Conclusion

Throughout several performances and experiments, several artistic goals were achieved, such as autonomy of music in multimedia art, augmenting expressiveness of performance with visualization, musical aesthetics in visual representation, and broad applicability of audio-based visualization to new compositions. From the result of the survey, one concludes that audiences can perceive the congruencies from the performance, and audio-based visualization can influence the appreciation of live music performance. Moreover, the degree of the congruency can differ depending on musical background, especially on the length of musical training and on the specific implementation of audio-based visualization. Additionally, from these issues and values a more concrete future research can be derived, including a compositional study combining live and pre-recorded video, various visualization methods using different types of cameras and computationally repositioning of each view, or a composition for theme and variations for diverse visual representations, etc. Additionally several psychological studies relating to multimedia art are possible.

The development of multimedia art will continue to evolve and transform into new ideas. It is the responsibility of the composer to lead the evolution of integrating sound in multimedia art to expand the boundaries and meaning of art. The rapid development of technology was caused an evaluation of the definition of the term multimedia. Powerful mobile devices and small embedded computing devices are ubiquitous, and the usage of audio-visual implements has proliferated. New applications of audio-visual equipment can be found in many contexts, and the complexity of its usage is decreasing. Composers must face this reality and develop new methodologies along with the evolving meaning of multimedia. The author believes the methodology presented will contribute not only to augment the expressivity of the performer in multimedia art, but also to inspire other composers who are defining these multimedia art forms.

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[Abstract in Korean | 국문 요약]
음악연주 시 표현적인 신체움직임의 오디오기반 시각화: 세 전자음악작품의 방법론 평가

오예민

시각 예술가와 행위연주performance 예술가, 음악가, 프로그래머들 사이의 합동 작업이 증가하면서 멀티미디어 행위연주 예술에 대한 탐구도 많아지고 있다. 오디오기반 시각화audio-based visualization의 방법은 소리 정보를 시각화된 신체적 표현과 통합시켜, 행위연주의 표현성을 높이는 것이다. 소리가 비디오 프로세싱을 제어하고 조절하게 함으로써 음악을 잘 나타내는 데 중점을 두었으며, 반면 영상은 소리에 어떠한 영향도 끼치지 않는다. 즉, 소리는 영상으로부터 독립적이다. 오디오기반 시각화는 청중에게 보다 깊게 음악을 감상할 기회를 줄 것이다. 이러한 특유의 방법을 활용하여 세 작품이 만들어졌다. 각 작품에 대한 경험 설문 조사를 통해 기술적이고 미학적인 가치를 평가함으로써 작품 구현의 실상을 질적으로 분석하였다.

Multicultural Music: Persian Electronic Music

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This paper will present the concept of "Multicultural Music" with a focus on “Persian Electronic Music”. By investigating local and global aspects of music traditions, the author has developed the idea of "local and global electronic music". There is a short introduction to traditional Persian music, and the Persian instrument *Santur* including its characteristics, such as the instrument's physical structure, possible performance techniques and limitations. These themes are discussed in regard to the composition of contemporary music and computer music in particular. Finally, some compositional tools and methods will be proposed and the artistic applications thereof will be discussed using "Le Corde Sul Corpo" as an example. Technical topics include an equation similar to McAdams equation, modal synthesis based on filters realized in Max/MSP and the detection of partials and harmonies in Open Music will be mentioned.

After trying to use elements of Persian music in electroacoustic compositions I decided to continue to explore the use of material from different musical cultures (especially Persian music) within my approach to the composition of electronic and computer music, as well as the involved tooling.

The phenomenon of multiculturalism is not often taken into consideration because of nationalistic attitudes; "You can look at the world from a national perspective, but you can lose yourself in a multicultural world remaining tied to nationalistic identity” according to Alireza Mashayekhi (Gluck 2006). Although the national identity influences personal identity, both are not necessarily the same.

While studying at the conservatory of Rome I met a French musician with whom I later composed the piece “*Polvere della tor di Babele*”. Communicating mostly in Italian, we started to work with long discussions about our cultures and languages and we tried to compose the piece using mostly our native languages (Persian and French). We had combined different musical languages and different musical material through electronic music techniques. This collaboration triggered mutual reflections about various aspects of multiculturality.

Local and global Electronic Music

Later while I was working on the third part of Unknown project, which became "*Return*" (for Santur, Saxophone, and Electronics), I asked my self the question: Could this be considered "Persian Electronic Music"? I.e. is it Persian music realized through electroacoustic techniques or is it electronic music with various sound objects, in particular Persian instruments? This question inspired my interested in the idea of “global and local electroacoustic music.

Sometimes I have traditional instruments play melodic material that distinctly evokes the Persian tradition and its atmosphere. I call this Local Electronic Music: The material draws from a precise tradition. At other times I approach the instruments from a purely timbric point of view, without any intention to make geographical or cultural references. The role of the electronics is very important to connect and unify the instruments and/or musical cultures.

Probably in a few years, in the field of electronic music, we will be able to find lines of domestic production (membership of a particular school or environment): e.g. electro-acoustic music of India, China or Latin America; after some electronic music festivals, I was able to identify styles of composition related to specific countries, just as it was possible to distinguished the French school from the German and Italian in the past: global electronic music on the other hand could still make use of traditional instruments and timbres, but it would not be utilized to evoke cultural connotations and would not contain any geographical or cultural references.

Iranian pioneers of Electronic Music

I think it is necessary to introduce two influential Iranian pioneers of electroacoustic music. The first, Alireza Mashayekhi, composed many works that are concerned with Multicultural music. For instance, his piece Shur Op. 15 was published on the vinyl “Electronic Panorama” in

1970, along with some of the most renowned, international pioneers of electronic music. The way applied the elements of Persian traditional music through electroacoustic techniques and the dialog (contrast) of these elements and noise material is very remarkable (Global Electronic Music). The other pioneer of Persian Electronic Music, Dariush Dolat-Shahi, composed very interesting Electronic music pieces in the mid 1980's. The combination of Persian traditional instrument (*Tar* and *Sehtar*), synthesizers and recorded sounds (mostly landscapes) in his works evoke clearly geographical and cultural references. (Local Electronic Music).

The 20th century has seen many composers using material from Asian or African traditional music. I think there are a lot of composers in the current generation who come to Europe from different backgrounds of musical education to learn about and improve their knowledge of Western music. During my studies I was surprised to find that many of my fellow international students were mostly interested in creating typical Western music, rather than allowing their personal musical identity to be enriched by Western perspectives. This observation leads me to research this topic in further detail and see compare it to the experience of other composers. (For this purpose I plan to conduct some workshops/laboratories of Multicultural music in the near future.)

I believe that, in order to combine different musical cultures and create a Multicultural music, it is necessary to have good knowledge of the cultures. Arguably, the deeper a composer's understanding of cultures, instruments and materials, the more layers of multiculturality can be achieved in a composition.

An introduction to Persian music

In the following paragraphs I will establish some basic background knowledge on Persian music that is relevant to the topic.

Forms in Persian music

Persian music is often divided into urban music (music of the big cities) and ethnic music (the music of the different ethnic groups that live in smaller towns, villages and mountain areas) (Farhat 1990). Ethnic music has been passed down orally and is closer to the ancient tradition: both use the same intervals and ways are based largely on improvisation. In urban music there are three instrumental forms, *Pishdaràmad* (overture), *Chàhàrmezrab* (four plectra) and *Reng* (dance), and a vocal form, which is called *Tasnif* (ballad). The *Pishdaràmad* was invented by a master of the Tar, *Darvish Khan*, and used as a prelude to a *Daràmad*. The Pishdaràmad, which is the opening section of a Dastgàh, can be 2, 3 or 4 parts and its melody comes from Gusheh, which are important traditional melodies.

The Chàhàrmezrab is a solo piece with fast tempo and usually is based on the melody that preceded it. The third form is the instrumental Reng, a simple dance generally played in concluding the Dastgàh. The vocal form, the Tasnif, has a similar structure to Pishdaràmad and usually precedes the Reng. Persian music is mainly a unison in which the instruments of the ensemble playing the melodic pattern and is heavily based on improvisation.

Rhythm

To study the rhythmic aspect of Persian music one must take into account the pattern and the rhythmic structures of poetry and music written and improvised in Persia. From the standpoint of rhythmic music it includes songs from the Persian urban free rhythm, the Avaz, or rhythmic songs typically in 2/4 or 4/4 6/8. By contrast, 5/8 7/8 and 16/8 are more frequently found in ethnic music (Azadehfar 2006).

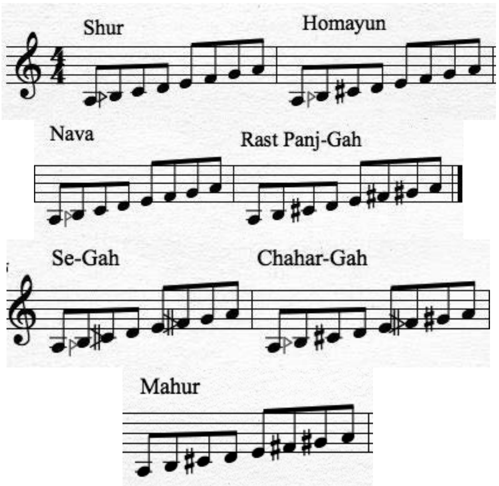
Persian Intervals

The system most widely used by musicians is Vaziri (Farhat 1990) whereby, in analogy with the tempered Western system, the octave is divided into 24 quarters of tone. For quartertone notations, two symbols are defined: *Sori* (𝇩) means 1 quartertone up and *Koron* (𝇪) means 1 quartertone down. An alternative notation has been proposed by Farhat using different intervals, including quartertone, semitone, whole tone and a greater range of tones (Farhat 1990; Heydarian 2005).

The Dastgah system

The *Dastgāh* is a modal system of traditional Persian music. A Dastgah is a melodic figure base with which the performer composes their pieces extemporaneously. The term Dastgah is often compared to the musical modes of Western tradition, although this is not the correct meaning of the term. We can describe it in this way: the Dastgah is the name of the initial mode of a piece of music and the song which comes back again and again, but mostly it is the name of a group of modes grouped according to tradition. Thus, the Dastgah defines both the name of a group of modes and the name of the initial mode of a piece of music. The etymology of the word comes from Dastgah: "The position (gah) Hand (dast) on the neck of the instrument." The term Dastgah represents a system, and is therefore a set of discrete elements, heterogeneously arranged from a hierarchy - coherent but still flexible. Although there are more than 50 modal systems, most theory is concerned with the 12 Dastgah: *Dastgah-e Shur, Avaz-e Abuu Ata, Avaz-e Bayat-e Tork, Afshari, Avaz-e Dashti, Dastgah*-e *Homayun, Avaz-e Bayat-e Esfehan, Dastgah-e Segah, Dastgah*-e *Chahargah, Dastgah-e Mahur, Dastgah-e Rast Panjgah,* and *Dastgah*-e *Nava* (Farhat 1990; Miller 1999; During 2006).

In conventional classifications, Abu-ata, Dashti, Afshari and Bayat-e-tork are considered subclasses of Dastgah Shur. Even the Bayat-and-esfahan is defined as a subclass of Homayoon, thus reducing the number of main dastgāh to a total of 7.



The subclasses are conventionally called "Avaz". Each Dastgah possesses a tonal center called Shahed. Each Dastgah has a number of derivatives called Gousheh.

The performance in each Dastgah begins with a section called Opening Daramad followed by modulations to other modes, the Gushe, during which the Shahed gradually moves upward, the execution ends with a sentence that leads cadence called Forud the initial mode of Dastgah (Farhat 1990; During 2006).

These are the sequences of intervals comprising the Dastgah:

Dastgah	Second	Third	Fourth	Fifth	Sixth	Seventh	Octave
Shur/Nava	3/4tone	1.5	2.5	3.5	4	5	6
Homayun	3/4	2	2.5	3.5	4	5	6
Mahur/ Rastpanjgah	1	2	2.5	3.5	4.5	5.5	6
Segah	1	1 3/4	2.5	3.5	4 1/4	5	6
Chahrgah	3/4	2	2.5	3.5	4 1/4	5.5	6

Radif (row, series)

> The Radif is the principal emblem and the heart of Persian music, a form of art as quintessentially Persian as that nation's fine carpet and exquisite miniatures. (Nettl 1987)

The Radif (meaning "order" in Persian) is a collection of melodic figures, broken down by the Dastgah, and passed down orally. Over the course of generations, the personal interpretation of each teacher has created new melodies that were added to the collection. The conservation of these melodies depends mainly on memory and mastery of each successive generation. To really learn and absorb the essence of Radif requires several years of practice and repetition; a master must internalize the Radif to the point of being able to perform any part at any time. A Dastgah portrays a specific sonic space and can hold ten to thirty *Gousheh* (corner, section, piece). The Gousheh of Dastgah may have different scales contained in Dastgah. The Radif for Tar are one of the most famous collections of Radif, a collection of ancient melodies that includes 20 to 40 per Gousheh Dastgah. The *Radif Mirza Hossien Gholi* Radif is the oldest still in use. The Radif consists essentially of parts not measured (free time) that provide a generative model or pattern for the creation of new compositions from the measured time but also of free improvisation. The Radif is a musical treasure of great wealth, the study of which can be approached from different angles: the theory, practice, education and cultural sociology.

Example 1. Dastgah-e Shur, Goshe-e Daramade-e aval, from Radif Of Mirza Abodllah (During 2006).

A brief history of Santur

Santur appears for the first time in *Moruj Al-Zahab*, a history book written by *Abol Hassan Ali Ibn Hussein Masudi* (X century), which mentions the Persian Santur about the music and its various instrument during the Sassanid Empire. And later we see other references to this instrument in the poems, books and etc. Since the 19th century the Santur has appeared in the shape that we know it today.

This instrument with small differences in shape and name appears in various geographic areas. In China is called *yang Quin*, in India *Santoor*, in Greece *Santuri*, in

Italy *Saltario*, in Germany *Hackbrett* and etc.

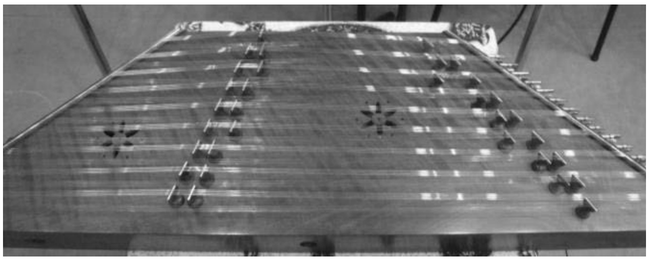


Photo 1. Persian Santur with double line bridge.

Tuning of Santur

A Santur with 9 bridges has 72 strings, each with its own pin and key tuning. The Santur has an extension of three octaves (E3 F6). Each musical note is given by a bundle of strings tuned to the same pitch; each bundle of strings rests on its own bridge (Kharak). The Santur with two rows of nine Kharak (18 bridges in total) is referred to as "Santur nine Kharak". The Santur is tuned using a special key, and its tuning, presents difficulties similar to tuning a piano. Compared to other Persian instruments Santur has some limitations and it is not possible to plan all of the Persian modes without retuning it. In some cases the musician changes the position of Kharak to tune the strings (Naini).

Photo 2. The right side of Persian Santur, with the keys tuning.

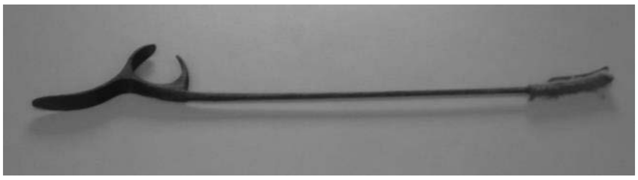


Photo 3. Mezrab (Santur's stick).

Software

Open Music

Using the General function, INHARM-SER object inside the software Open Music, it is possible to generate a list of partials based on initial frequency (f0), the coefficient of distortion and the number of partials.

Example 2. First 12 partial of the note C2 (65Hz) and distortion coefficient of 1.43.

Max/MSP

To realize the Live Electronics part of my piece "Le Corde Sul Corpo" I used the software Max/MSP.
The sound of a Santur was recorded using two microphones at a distance of 18-21 centimeters above the bridge (Kharak). Converted into a digital signal from an audio interface, and routed into Max/MSP. Using a Matrix we can get different routing of microphones and various tools used (sub patches): We have 6 frequency shifters, 8 delay lines, 4 Modalyreza (filter bank) and Spat (spatializer by IRCAM).
The outputs of the processes from the matrix in 8 sound sources are sent to a spatializer with 8 output channels. After trying different methods for real-time spatialization, I decided to use Spat for this piece, due to its flexibility. For instance, Spat makes it easy to vary the number of output channels, which comes in handy when playing in different concert halls with a different number of speakers.

Modalyreza

Modalyreza is a patch made in Max/MSP. It implements a virtual resonant body simulating the resonance modes. It uses a bank of band pass filters whose center frequencies are resonant frequencies of the resonance modes of the virtual body.

Using as center frequencies the values of a mathematical series can generate different spectral families. Two mathematical series are particularly interesting musically: the harmonic series, by the timbre of many instruments, and the geometric series, with a ratio twelfth root of two, from the distribution of the frequencies of the tempered system. Using the generalization of the harmonic series and geometric series we can obtain several families of spectra.

Our spectrum is then described by the equation:

> $F(n) = f(0) * (n \wedge \alpha * \beta \wedge n + \gamma)$ n is the order of the partial and coefficients α, β and γ are real numbers (Cella 1976).

Alpha is the exponent of "expansion / contraction" of the spectrum:

> If $\alpha = 1, \beta = 1, \gamma = 0$ the series is harmonic if $\alpha > 1, \beta = 1, \gamma = 0$ the harmonic series is expanded, if $\alpha < 1, \beta = 1, \gamma = 0$ the harmonic series is compressed beta is the basis that deter-

mines the "relationship" between a component of the spectrum and the next.

If α = 0, β> 1, γ = 0 the series is geometric.

So in this case if β = 1 the ratio is the unison.

If β = 2 the ratio is the octave.

If β = 12 root of 2, semitone.

If β = 24 root of 2, 1/4 tone etc.

For each value of α if β ≠ 1 γ = 0 the harmonic series is deformed geometrically.

For each value of β if α ≠ 1 γ = 0 the geometric series is deformed harmoniously.

The coefficient γ determines a shift frequency, f (0) * γ is a constant which is added to the whole series for each value of α and β if γ ≠ 0 is applied to a translation in all the series.

So if α = 1, β = 1, γ = 1 we have a harmonic series, but starting from the second harmonic.

If α = 1, β = 1, γ = 3 we have a harmonic series, but starting from the fourth harmonic.

To control the Q of the resonant filters we use this other equation.

Tau = Q / n ^ damping.

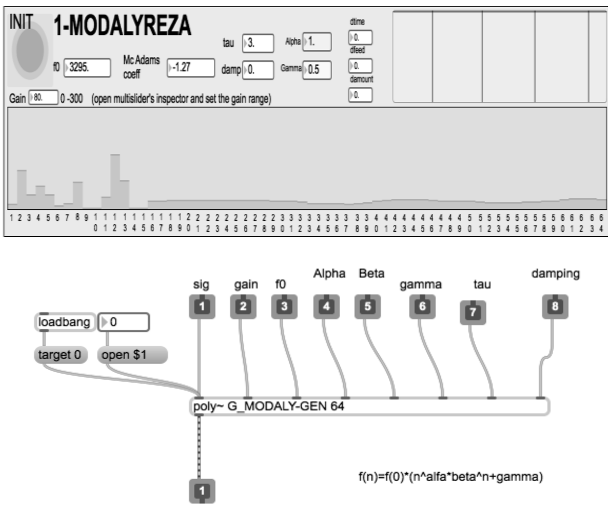


Figure 1. MODALYREZA, has used the *poly~* in MAX/MSP.

Goals and realization

In my music you can clearly perceive the movements of the masses and layers of sounds. When these sound masses come into collision you will have the sensation that occur phenomena of penetration or repulsion, and that certain transmutations that occur on certain plans are projected onto the other, which move at different speeds and in different directions. There will be not more place for the old conception of melody or combination of melodies: the entire work becomes a “melodic totality”; the entire work will flow like a river. (Varèse 1976)

Santur and new music

As mentioned above the Santur is an ancient instrument; in Iran the instruments (and music in general) are closely related to the tradition, a long-time evolution of the instruments and Persian music is almost stationary, some instruments have been developed further, but without great success. A major problem with Santur is that it needs to be tuned each time for each Dastgah or scale that you want to play. In general, the tuning of Santur is a very difficult operation. And this operation takes time.

For example, saying in a Western way, if you play the major scale and you want to change in minor, you need to change the pitch of the strings. Another limitation is the changes of position of the hand on the instrument. For instance, playing in G is easy, as the relevant notes are disposed next to each other and there are no notes too far apart. But the same Dastgah in C becomes much more difficult, requiring many jumps between yellow strings and white strings (left and right side of the instrument, tuned with an octave of difference between them). A much faster solution to change the pitch is to double the bridges. I think doubling the bridges will have many opportunities to tune the instrument and experiment with nontraditional tunings and articulations as the glissando performed by varying the position of the bridge.

For example in Santur with a single bridge (Kharak) it is not possible to have G and G# in the same position (in the same octave). For Persian music this doesn’t matter, if we want to play in D Shur we will need the D and the D Koron (¼ tone below) but they cannot be in the same position on Santur, also according to traditional Persian music these notes are not played one after another (chromatically), so it's not a problem for traditional music but it is to play other kinds of music that require different sets of notes.

I therefore find it very useful to duplicate the bridges of the Santur, allowing the expert instrumentalist to change the pitch of each string quickly. I think the simplest thing is tuning the strings a tone below and then the musician can move ¼, ½ or ¾ tone above. This slightly varies the timbre of Santur because of physical changes in the instrument, such as string tension, the pressure of the bridges on the body and the location of contact points. This method comes at a slight cost regarding the brightness of the sound.

However, the increased flexibility is often more important.

The relationship between instrumentalist and composer

I decided to write “Le Corde Sul Corpo” (Ostovar 2014) when I met Kioomars Mosayyebi, a very experienced Santur player. After speaking to him about the idea of a piece for Santur and electronics, we had a long discussion about contemporary music and electronic music. We reflected on what could be done by starting from the traditional Persian music and he has proved very interested in collaborating and playing this piece.

It was stimulating for me that he was very open to different types of music and had played a lot of World Music, Jazz and Blues with different formations. Very often the players of Persian traditional music are not interested in playing other kinds of music, in particularly, contemporary music, which is unknown to many of them. I think it's very important that the performer understands the music that he or she is playing in a way that he/she can perform to the best.

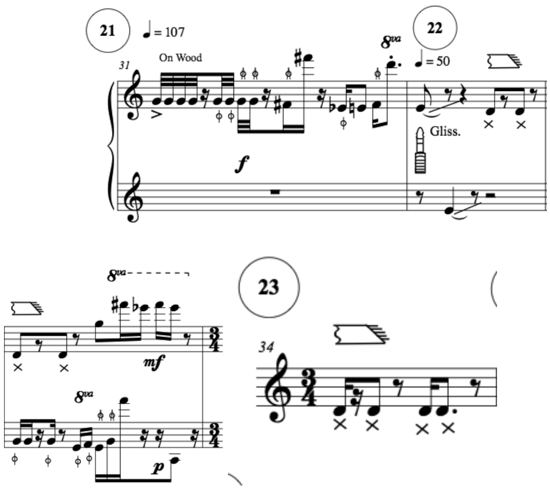
Talking to him about this experience he told me he also felt the need for a change and to have something new in Persian music. He found it very interesting to play contemporary music and listen to new sounds from his instrument, playing it in ways that he had never tried to play it; it was also hard to see his instrument in this way. Also working with "live electronics", hearing the electronic sound in real time was a new experience of musical performance and during rehearsals he had the chance to learn a lot.

The sound of the Santur

In the piece “le Corde Sul Corpo”, the melody is not important, unlike in traditional Persian music. I actually tried to eliminate the melody, keeping the sound of the instrument and trying to find the colors and sounds that I wanted. I think one of the problems when writing contemporary music for Persian instruments is to hide the scales and intervals of this music, because they very quickly resemble traditional music very, or otherwise we must think about how we want to use them. In this case I decided to avoid using the intervals from the traditional Persian music and the repertoire of Santur.

Preparing the Santur allowed me to overcome the limitation to fixed pitches and also to dive deep into the exploration of new timbres of the instrument. Preparing Santur and taking the sound from other parts of the instrument gave us the possibility to transform the sound of the instrument before it goes to live electronics. Many of the techniques found and developed by us during the research period and rehearsals, require practice and refinement of instrumental techniques and confront the performer with new problems. For example, to get the sound by touching the keys of the Santur it is important to understand how to change the position of the hand, because hands are holding the Mezrab (stick) and the performer should play the keys with his hand and then back on the strings and play with the Mezrab (Bars 32 and 34). To my knowledge, this technique has never been used in traditional Persian music.

Photo 4. Prepared Santur for the piece “le corde sul corpo”.



Example 3. Bars 32-34, score of the piece “Le Corde Sul Corpo”.

Rhythm

I find it very interesting to work with Persian rhythmic patterns. These are very rich, but in this piece there is any particular rhythmical pattern taken from traditional music. I started from very short gestures and gradually moved to longer and more complex gestures until the section (bars 58 to 88) formed by a long gesture made of tremolos and trills (Riz: a traditional technique of Santur).

Dynamic and Density

Typically, traditional Persian music is notated without dynamics (see the examples of Radif above). This leaves a lot of freedom to the performer to improvise. By contrast, in contemporary music and especially in the electronics, the composer tries to control and choose very precisely each parameter and especially the dynamics.

In the piece "Le Corde Sul Corpo" I paid much attention to the density and dynamics. I used a rather large range of dynamics, from PP to FFF, larger than those typically used by Persian music. Finding different dynamics in the same bar is a new situation for a Santur player. Obtaining a large dynamic range requires more control of touch. The density parameter is structured around the score and is expanded by the live electronics. For example in the section of Riz (Tremolo and Trilli, bars 58 to 88), the goal was to have a unitary dynamic (balanced) and form a mass of sound by adding short sounds, changing the timbre of the sound several times and generating a large "gesture-texture". When the performer plays very fast (trying to maintain a constant dynamic in the whole part, even though he is playing different parts of the instrument that respond differently to the percussion of the Mezrab) sometimes the result was not what I wished but here live electronics really helped to overcome these limitations and expand the possibilities of the instrument.

For example in the section of the Riz, I utilized:

- The Delay for adding the density.
- The Frequency shifter to extend the frequency range of the instrument.
- Modalyreza to sculpt further altering the tone and color of the instrument's sound, to lengthen the duration of the sounds and to get the timbre of electronic sounds.

Conclusion

In this article, I focused on Persian Electronic Music. By sharing my experiences especially in a piece for a specific instrument (Santur and live electronics) I tried to give a accurate example of the process from traditional music to contemporary music.

By crossing different musical culture and considering the electroacoustic music and traditional music from a multi-cultural perspective we could find a new perception in the music.

I know there are many aspects to develop to go deeper in this topic and discover new possibilities with different traditional instruments from different cultures. I hope in future we can see more examples and researches in this subject.

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[Abstract in Korean | 국문 요약]
다문화 음악: 페르시아의 전자 음악

알리 오스토바

이 글은 "페르시아의 전자 음악"을 중심으로 "다문화 음악"의 개념을 제시한다. 저자는 음악적 전통을 지방특유의 관점과 전세계적인 관점에서 보고, "현지의, 전세계의 전자 음악"에 대한 견해를 발전시킨다. 페르시안의 전통음악에 대한 간단한 소개와 함께 페르시아 악기, 산투르*Santur*의 악기구조, 가능한 연주 기술과 한계점 등, 그 악기에 대한 특징들을 설명한다. 이 설명은 특히 현대음악과 컴퓨터음악을 작곡하는 시점으로 논의된다. 끝으로, 몇몇 작곡 도구와 방법을 제안하고, 그 도구와 방법이 예술적으로 적용된 예시로서 《몸체위의 현Le Corde Sul Corpo》을 논한다. 기술적인 부분으로 맥아담스McAdams 방정식과 유사한 방정식, 맥스/엠에스피Mas/MSP로 만들어진 필터filter 기반의 모드 합성Modal Synthesis, 그리고 오픈뮤직Open Music을 사용한 배음과 화음 탐지 장치를 언급한다.

The Problem with Pepper's Ghost: Incorporating Pseudo-holographic and holophonic technology into the contemporary music performance space

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With the emergence of holographic projection technology we have witnessed the rise of the phenomenon of the holographic performer. This article examines the relevance of the holographic performer and how this trend might influence the future of live music performance. The research looks at some of the developing technologies based around holographic products and techniques, and seeks to implement them into a novel performance space. This space is designed with the intention of further exploring an emerging performance paradigm that integrates holographic techniques with live musical performance. To execute this project, I have worked with industry technology, and tested some self designed technology solutions to create a unique 3D holographic performance space. This performance space is designed to be scalable to suite venues ranging from mid-sized theatres to major music festivals and hopes to inspire reimagining's of the conventional contemporary music performance environments.

The performance technology landscape grows and evolves at such a rate that the contemporary music performance arena finds itself faced with an endless range of new tools to extend it's communicational reach. With the exponential power growth and decreasing cost of computing, technologies like projection mapping and motion tracking are becoming regular features of large-scale festival and arena spectaculars as well as smaller scale performance. One of the more fascinating and potentially far-reaching facets in this emerging techno-entertainment landscape is that of holographic technology.

The holographic performer

The emergence of holographic performers

In 2012, the Coachella music festival introduced the world to the idea of holographic performance when Rolling Stone hall of famer Tupac Shakur (Edwards 2009) performed a set with renowned artist and rapper, Snoop Dogg. Shakur had been killed in a 1996 shooting, a good 16 years prior to his partnering with Snoop on the stage in California. This posthumous appearance was received with mixed reviews but critically it demonstrated to the world the potential powers of holographic performers (Wain 2014). For a short time the music industry invested in creating resurrection based holographic performances. Michael Jackson performed his hit "Slave to the Rhythm" at the Billboard Music Awards in 2014, five years after his death, leaving fans questioning the ethics, taste and validity of the use of this kind of holographic technique (Wain 2014).

In fact, this pseudo-holographic illusion has been used for centuries and is commonly incorporated in theatre performance (Naughton 2006). In 2006, English Virtual band had used the technique to perform at the Grammy awards alongside Madonna (Grammy 2015). However, the Coachella performance was significant as it was the first use of the technique for a solo artist.

Recently, holograms have even had shows cancelled by law enforcement due to fear of them causing civil unrest. In 2015, a holographic appearance in Chicago by rapper Chief Keef was cancelled after Mayor Rahm Emanuel's office stated the artist was "an unacceptable role model" who's holographic performance could pose a "significant public safety risk" (Byford 2015).

Each of these performances marks significant points in the development of the holographic performer. In each case they have created controversy and inspired meaningful questions based on appropriation, ethics, civil unrest and censorship. Considering two of the three aforementioned artists did this from beyond the grave, the outcomes of such performances certainly attests the power of this emerging medium.

Contemporary settings

Holographic events are now commonplace in the high end of the entertainment sector. (Musion 2015). In South Korea's capital Seoul, a significant investment in this technology has taken place. "KLive is the world's first dedicated Kpop performance hall" reads the blurb of the online biography (Klive 2015). KLive is a dedicated and purpose built holographic venue, boasting a 270-degree immersive holographic experience. Some of Kpop's biggest stars have either developed shows or are debuting shows in the near future. In 2015, in this venue, major Kpop boy band G-Dragon premiered their holographic show. International hip-hop and Youtube sensation Psy plays on a bill alongside hit-makers 2NE1 and Big Bang two times a day almost every day of the year.

This kind of concert schedule is unheard of outside of Las Vegas, Broadway or the West End. So how is such a demanding performance run possible? The nature of these holographic performances means the members of these top-ranking acts don't need to be present at the venue. In fact, shows are pre-recorded and projected onto a screen, played in a similar manner to a movie. This poses a question, does it become hard to separate the realm of the dedicated holographic venue from the normality of the 3D cinema. Has the holographic performer become the 21st century equivalent of lip-syncing?

In Japan, the holographic performer has been taken to a new level. Hatsune Miko is not what some might call a real person. Her makers describe her as "digitally synthesised voice encapsulated in a crowd-sourced humanoid persona" (Behrens 2014). Miko has over 1.8 million Facebook followers and in excess of one hundred thousand original songs in her catalogue. She plays sold out concerts all over the country with a live band, to crowds of crazed fans desperate to see her dance and sing. When you consider she is only a digitally animated projection, this seems almost surreal. Journalist Sam Behrens describes the phenomenon as "a projection of national preferences" and goes on to state he feels "her impact is real, even though she is, arguably, not" (Behrens 2014).

Carbon copy, cheap fake or something else

Hatsune Miko is a handcrafted and crowd sourced persona based on mass marketing research delivery. She is only manifest in hologram form and exists, arguably again, in her own right. Meanwhile the other previous examples are complex pre-recorded reproductions. This being the case, does the reproduction of a real person have a sustainable market in a world often governed by perceived authenticity? The basest definition of authenticity questions if a work of art is an original or a forgery (Guter 2010). In this sense a holographic performer can only be the latter, but what of Heideggars authenticity of perception and self (Guter 2010)? The existing incarnation of the hologram has attempted to present itself as live performance or theatre. McConachie believes that "good theatre" involves the exchange of "emotionally charged entanglements of truth" between audience and players. He goes on to say "this engagement happens among live participants in the same space and during the same time" (McConochie 2011). If the key to McConochie's "good theatre" is interaction in the same space at the same time, one could argue that the holographic performer somewhat fits the criteria, but in terms of all participants being live, or even alive, the hologram falls short. Until computational intelligence reaches a point where it can pass some kind of performance Turing test, it seems the holographic performer won't compete with the performance of living musicians and artists in the same space as their audience. It would seem for now the holographic performer works most effectively when achieving the impossible, like raising a star from the dead or achieving a feat of illusion. Is this simply a gimmick designed to play into publicities hands?

The AED defines gimmick as "a trick or device, designed especially to attract attention, publicity, or trade". Delving further into the etymology of the term Brewster defines "gimmick" as used "to describe some device by which a conjurer or fairground showman worked his trick." If we take this as our guideline then the glove appears to fit in most cases.

If the holographic performer, for the lager part, is simply a gimmick, then what happens when the technology become as available as it's cousins, motion capture and projection mapping? Will the holographic performer become a thing of the past looked at as a cringe-worthy throwback from yesteryear? In an attempt to find an answer to this I first examined the history of the technology and then set about designing a solution based on my findings.

Beyond the holographic performer

The Research Problem

Is the holographic performer a gimmick with limited lifespan, and if so, how will holographic technology play a part in the future of music performance spaces?

The ghost of holograms past

By now, those with any knowledge of this medium will be aware that all of the examples of so-called holograms used in this article so far are technically nothing of the sort. In fact they are a high-tech versions of an illusion that was first describe in 1584 by Neapolitan scientist and scholar Giambattista della Porta. In his work Magia Naturalis or Natural Magic, he titles the description of this effect as "How we may see things in a Chamber that are not" (Naughton 2006).

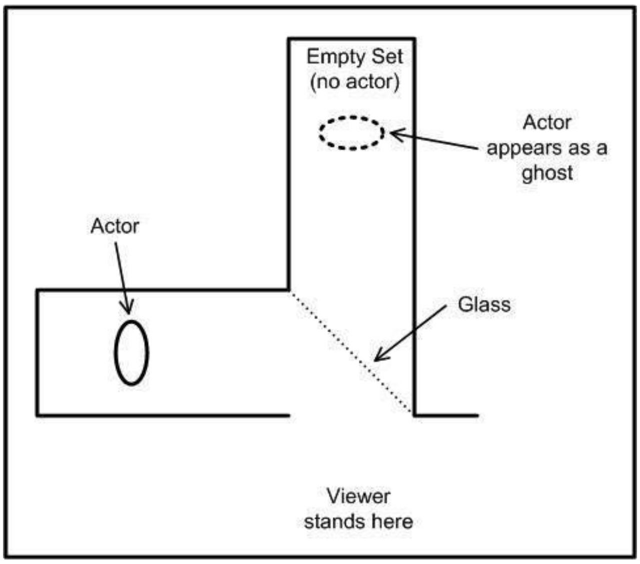

Figure 1. Visualisation of Porta's description.

In 1862, British civil engineer Henry Dricks set about trying to discredit the charlatanry of the popular spiritualist and phantasmagoria movement. He was frustrated with the use of magic lanterns and other gimmicks in this medium and developed a small contraption designed to demonstrate the power of science over superstition. This contraption (fig. 1) used a sheet of glass and well designed lighting to project an image on an actor onto stage (Harries 2000). The year following, the contraption and the associated illusion was brought to popular attention when John Henry Pepper, a chemistry professor at London Polytechnic Institute, built an operational full-sized version and exhibiting it (Naughton 2006). This demonstration led to the illusion and its consequent variations being referred to as 'Pepper's Ghost".

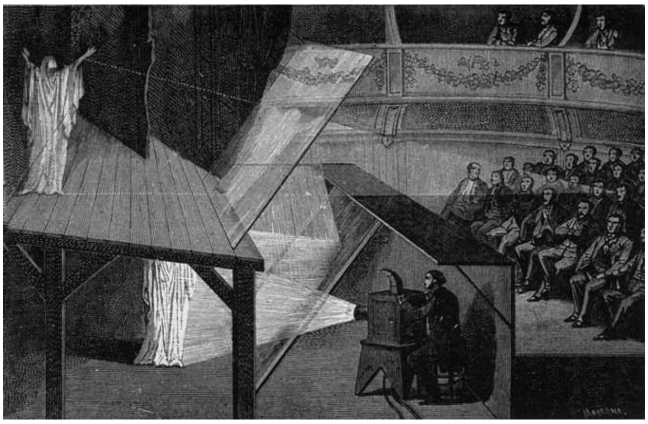

Figure 2. Dramatization of Drick's contraption.

The so-called holograms that graced the stage at Coachella and the Billboard music award are more correctly advanced interpretations of the Pepper's Ghost conjuring illusion. This historic lineage does not make the illusion any less impressive, especially with the recent significant growth in supporting technologies such as projection mapping, laser projection, Holographic Eyeliner foil and motion tracking.

Moving away from the ghost

As I discussed earlier, the holographic performer holds promise as a stand-alone entertainment format, but how do we focus attention on incorporating this type of lighting illusion into contemporary live music performance? Instead of focusing the attention on the illusion of the holographic performer, the potential to integrate this technology into live performance is boundless. As an alternative to holograms performing on real stages, why not approach the technology from the other direction and examine the possibilities of live performers in holographic environments. In this scenario the performers would both interact with, and integrate themselves into holographic lighting and staging. This approach could place the performer into environments only limited by venue size and imagination. In addition to using the Pepper's Ghost illusion, incorporating other existing technologies into this space has created spectacular results. There are numerous creative collaborations around the world exploring the potential of combining these technologies.

Motion Tracking

In the last decade, motion capture and tracking technology has moved out of the production studio and into the home. All major gaming platforms have implemented the technology in some format to make home gaming more physically interactive. The spread in available technology has lead to it motion tracking being integrated into both large and small-scale performances. Nowhere is this more pronounced than in the world of contemporary dance.

French dance project Pixel describes itself as "a dance show for 11 dancers in a virtual and living visual environment. A work on illusion combining energy and poetry, fiction and technical achievement, hip hop and circus. A show at the crossroads of arts…" (Pixel 2014). In this show dancers interact with rear and down-projected matrices that create a plethora of virtual spaces for the players to perform within. The visual effects include virtual rain made of light, holographic hoola-hoops, and scrolling mountaintops that threaten to swallow up the performers. It appears these projected elements react to the movement of the performers by tracking the their motion, it is not clear how this motion tracking is executed but the effect is mesmerizing.

Projection Mapping

The incorporation of high lumen projectors into Pixel's performance made the effect all the more stunning. In a similar manner to the growth of motion tracking, projection technology has become more powerful and accessible. The introduction of laser projection has made the domestic projector capable of extremely high resolution with pinpoint accuracy (Jones 2014). This has resulted in a further development in the medium of projection mapping.

Examples of this technology date back as far as the early 1960's when the medium fell under it's more academic name, Spatial Augmented Reality. The technique was first used in a public display as part of Disneyland attraction the Haunted Mansion. In 1980, Artist Michael Naimark used a version of this technique in his installation Displacements. Used a rotating camera in a living space then simply replaced the camera with a projector showing the footage (Jones 2014).

In 1999, significant momentum gathered behind projection mapping as academia began to focus on the technology. The paper "Table-top spatially-augmented reality: bringing physical models to life with projection imagery" discussed the potential of the virtual office and how best we can incorporate projection mapping into our everyday lives. The research team behind this paper was responsible for significant growth in smart projection and how the modern world views projection technology (Raskar / Ramesh / Welch / Chen, 1999).

Before long both advertising and the arts began using the projection format. The Electronic Dance Music (EDM) movement became a champion for the technology by incorporating it into their live shows. The technique has also been used in public art installations in capitol cities around the world. Lyon's Fête des Lumières showcases the dutting edge of public projection mapping installations (Art Visuel 2014).

Hybridised Systems

In the contemporary creative landscape both projection mapping and motion tracking are commonplace. Due to the powerful computing solutions now available to the consumer market, some artists have been combining the two with extraordinary results. A collaboration between Japanese media artist Nobumichi Asai, makeup artist Hiroto Kuwahara and French image engineer Paul Lacroix has resulted in project Omote. Using a combination of projection mapping, specialist makeup and motion tracking techniques the trio project a series of modernist and surreal images on the face of a model. The result demonstrates the visual power and limitless imagination of combining these mediums (Omote 2014).

The immersive theatre

Since the combination of projection and tracking techniques delivers such affective and significant results, it begs the question how else can we combine available technology to enhance our performance potential? In an attempt to answer this question I have begun to develop a holographic environment in which a real or live artist will perform. This environment borrows element from the holographic world and incorporates them into the live performance realm along with a number of other developing technologies. Using a multi-layered technique I hope to create an illusion that appears to have no edges or limits. I scale model has been designed for testing on the concept. When the project is brought to full scale I will introduce the projection mapping, motion tracking and directional audio elements. The projected elements will be managed by Kayne Hunnam, the research's primary projection designer.

The hardware

To achieve this immersive virtually augmented performance space, IT is important we examine the available technologies and how they may be best applied.

Musion is the industry leaders in contemporary holographic projection foil technology. Their flagship product, Eyeliner Foil, is a polymer screen that plays the same role in contemporary holographic displays as the sheet of glass in the Pepper's Ghost illusion. This technology was used for the Tupac resurrection performance and is a commonly used medium for holographic projection.

By layering three sections of Eyeliner Foil one behind the other from stage up stage to down, three separate projectors will be used to create a front, middle and rear composite projection. The rear section will involve the use of rear projection to create a backdrop or rear effect layer that will provide the main body of the illusion. The Middle layer will use the Pepper's Ghost technique to project the main dynamic components of the performance. The front layer will use the same projection illusion to display other roaming and less focussed elements give the impression of depth to the stage configuration. The performers will be positioned in between these layers of Eyeliner Foil effectively playing within the hologram. Essentially, this technique replaces traditional stage lighting with high-powered projectors. Using this configuration the performers will be able to perform in virtually any space.

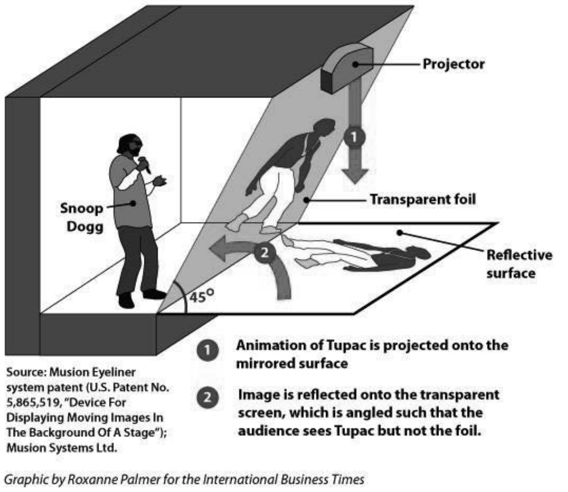

Figure 3. How the coachella illusion worked.

In order for the immersive theatre to be truly effective attention must also be paid to the audio element of a performance. Incorporating dynamic audio elements poses significant challenges when using the conventional Public Address (PA) system. The nature of sound is such that it only allows for high frequency content to by directional. The various hyper-directional audio solutions either only deal with high frequency content or use ultrasonic transducers that don't allow for high enough db (SPL) to be used in music performances. As a result, these hyper-directional systems are used for small-scale private audio installations like supermarket advertising or art gallery guides.

With the rising, and possibly peaked, popularity of what the US music market has deemed Electronic Dance Music or EDM(*) and wave table synthesis being readily available to any home producer, thick textured, multiple octave instruments are everywhere in music performance. These instruments are impacting and flexible. One of the most common uses of these instruments involves creating a LPF that quickly moves up the frequency spectrum to create what we know today as a riser.

Using this principle, I have developed a hyper directional speaker designed to rove over and audience providing additional high frequency content and creating an individual experience for every audience member. By adding moving sound objects within the conventional PA environment, similar to the concept of a roving stage light, a hyper-directional speaker would be able to emit beams of high frequency sound over on instruments low passed frequency content and create a new dynamic sonic element.

The Future of Integration

In this research, I explore the further combining of developing visual creative fields and directional dynamic audio elements with the intention of providing insight into the question: How will existing holographic and holophonic technologies help further the contemporary music performance space? Answering this question definitively is not possible as the music performance landscape is ever evolving to both meet and create demand. This paper has hopefully provided some insight into the background behind these techniques and technology. The research has been designed in an attempt to highlight and inspire innovative thinking as to how performers can integrate existing visual and sonic technology into their performance practice. Moving into the future I hope this article might provide some precursory stimulus in the on-going dialogue about music performance technology and the contemporary artist.

Acknowledgments

I would like to extend my thanks to Kayne Hunnam for his technical assistance and insight into holographic technology. I would also like to acknowledge Musion and their UK office for providing significant insight into the future of their technology.

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[Abstract in Korean | 국문 요약]

페퍼의 유령Pepper's Ghost의 문제점:
유사홀로그래픽Pseudo-holographic과 홀로그래픽holophonic 기술을 현대음악 연주공간으로 결합하기

앤드류 왈드

우리는 홀로그램 투사holographic projection 기술의 발명으로 인하여 홀로그램이 실연holographic performer되는 경이로운 현상을 보게 되었다. 이 글은 홀로그램 실연의 적절한 의의와 이러한 추세가 향후 라이브 음악 연주에 어떠한 영향을 미치는지를 연구한다. 이 연구에서 홀로그램의 결과물이나 기법과 관련한 몇몇 발전 기술들을 살펴보고, 이들을 참신한 연주 공간에 적용할 방도를 모색한다. 이 공간은 더욱이 홀로그램 기술을 라이브 음악 연주와 함께 융합하는 신흥 연주의 전형을 모색하기 위한 의도로 고안되었다. 프로젝트의 실행을 위해, 저자는 공업기술을 사용하여, 몇몇은 저자 자신의 설계기술이 쓰인 고유한 3차원 홀로그래픽 연주 공간을 만드는 실험을 하였다. 이 연주 공간은 중간 크기의 극장에서 큰 음악 축제까지 각 현장에 적당한 범위로 축소확장이 가능하게 설계되었고, 이 공간이 현대음악의 상투적인 연주 환경을 새롭게 재구상하는 데 감흥을 되길 바란다.

PART II: Reviews

제2부: 참관기

Looking Back, Looking Forward:
Review of *International Computer Music Conference 2015*

Kang, Joong Hoon

The 41st *International Computer Music Conference (ICMC),* since its first meetings in 1974, was in Denton, Texas from September 25 to October 1. This small city, with a population of about a hundred thousand, is located within an hour drive north-westwards from Dallas, and is known as a college city where *University of North Texas (UNT)* and *Texas Woman's University* are at the center of it. *UNT,* the old venue for the 7th conference in 1981, hosted this year again, and provided the chance to retrace the historical steps for decades of computer music and to find ways to discover new music for the future, with this year's subject, "Looking back, looking forward." There were three keynote speakers; Miller Puckett was the first speaker who talked about 'Recent and Upcoming Features in Pure Data,' a UK composer, Jonty Harrison addressed on 'Sound Diffusion Illustrated Demonstration' was another, and lastly 'What's New in Kyma7' was discussed by Carla Scaletti, the developer of Kyma System. There are about sixty papers represented in twenty-three paper-sessions in every morning for the total eight days. The noticeable topics are Interaction and Improvisation', 'New Interfaces for Musical Expression', 'Digital Signal Processing and Effects' and 'Music Information Retrieval and Perception,' among many.

뒤로 보고, 앞으로 보기:
국제컴퓨터음악학회2015 참관기

강중훈

1974년 첫 학회를 시작으로 올해로 마흔 한 번째를 맞이하는 국제컴퓨터음악학회International Computer Music Conference (ICMC)는 9월 25일부터 10월 1일까지 여드레 동안 미국 텍사스주의 덴튼Denton에서 개최되었다. 미국에서 아홉 번째로 큰 도시인 댈러스에서 북서쪽으로 차로 약 오십 분 거리에 위치한 덴튼은 인구 십만 명 정도의 작은 소도시이며 텍사스여자대학교Texas Woman's University와 이번 음악제가 열린 북텍사스대학University of North Texas (UNT)이 도시의 심장부를 이루는 대학 도시라 할 수 있다. 특히 북텍사스대학은 34년 전 1981년에도 일곱번 째 학회를 개최한 적이 있는 유서 깊은 장소로, 이번 학회의 주제 또한 지난 사십 여 년간의 컴퓨터 음악 발자취를 돌아보고 컴퓨터 음악이 앞으로 나아갈 미래의 비전을 모색하고자 하는 취지에서 "뒤로 보고, 앞으로 보기Looking Back, Looking Forward"였다.

이번 학회의 기조연설자는 모두 세 명으로 맥스/엠에스피Max/MSP와 퓨어데이터Pure Data의 개발자로 유명한 밀러 푸켓Miller Puckette이 '퓨터데이터에서 최근과 앞으로의 특성Recent and Upcoming Features in Pure Data'이란 주제로 첫 워크숍과 기조연설을 하였고 두 번째로는 영국 작곡가 존티 해리슨Jonty Harrison이 '소리확산의 예 제시Sound Diffusion Illustrated Demonstration'의 주제로 워크숍과 기조연설을 하였으며, 마지막으로는 키마 시스템Kyma System의 개발자인 칼라 스칼레티Carla Scaletti가 '키마7의 새로운 점What's New in Kyma 7'이라는 주제로 각각 워크숍과 기조연설을 진행하였다.

여드레의 학회기간 중 총 스물 세번의 페이퍼 세션이 오전에 열렸으며 여기서 발표된 약 육십여 편의 논문들 중 '상호작용과 즉흥Interaction and Improvisation', '음악적표현의 새 인터페이스New Interfaces for Musical Expression', '디지털시그날프로세싱과 이

펙트Digital Signal Processing and Effects' 그리고 '음악정보검색과 인지Music Information Retrieval and Perception'에 관한 내용들이 주요하게 발표되었다.

또한 이 기간 중 총 서른 한 번의 음악회가 자정을 넘긴 시각까지 매일 네 차례에 걸쳐 개최되었다. 먼저 '크로스루프CrossLoop'라는 음악회는 주로 테잎작품들을 위한 음악회로 오후 1시부터 4시 30분까지 1시간씩 총 세 번에 걸쳐 스카이극장Sky Theatre과 메릴엘리스 인터미디어극장Merrill Ellis Intermedia Theater, 두 개의 리스닝룸에서 번갈아 가며 열렸다.

오후 4시 반에 시작하는 두번째 연주회, 아쿠스매텍스Acousmatex는 북텍사스음악대학으로부터 1km정도 떨어진 캠퍼스 내 머치슨공연예술센터Murchison Performing Arts Center의 리릭극장Lyric Theatre에서 열렸는데 테잎작품과 라이브 일렉트로닉 작품들이 함께 연주되었다. 스펙트럼Spectrum이라 명명된 세번째 연주회는 저녁 8시 북텍사스음악대학 내 보에트만홀Voertman Hall에서 열렸으며 이 역시 테잎작품과 라이브 일렉트로닉 작품들로 구성된 음악회였다. 마지막으로는 밤 10시30분에 하루를 마감하는 음악회로 클럽일렉트로Club Electro가 라이브러리몰Library Mall과 리릭극장Lyric Theatre 그리고 캠퍼스로부터 약 3km 떨어진 러버글로브스튜디오Rubber Gloves Studio에서 번갈아 가며 열렸는데 이 클럽일렉트로는 격식있는 연주회라기보다 그날 하루의 뒤풀이 같은 캐주얼한 분위기로 약간의 알코올이 들어간 음료 등을 마시며 재즈나 팝, 아이디엠Intelligent Dance Music 등의 요소를 띈 작품이나 새로운 인터페이스를 사용하는 작품 등을 다 같이 즐기며 감상할 수 있는 음악회였다.

필자는 일정 관계로 첫째 날 늦게 도착하였기 때문에 이튿날 오전에 시작하는 밀러푸켓의 워크숍부터 참가할 수 있었다. 개인적으로 퓨어데이터를 사용자로서 소프트웨어의 최초 개발자가 진행하는 워크숍에 많은 기대를 했는데 실제로 푸켓이 다룬 내용은 상당한 흥미를 유발하는 것이었다. 필자가 그동안 퓨어데이터를 사용하면서 품어온 여러 의문점과 프로그램에 대한 몇 가지 희망사항을 이번 기회에 모두 꺼내지 못한 점은 아쉬웠지만, 개발자가 자신이 만든 소프트웨어를 직접 시연하고 설명하는 모습을 보는 것으로 무척이나 흥미로웠고 이로 인하여 퓨어데이터에 대해 좀 더 이해할 수 있게 되었다.

이번 학회는 총 서른 한 번의 음악회에서 약 삼백 여 편에 이르는 다양한 작품들이 발표되었는데 이렇게 많은 작품들을 모

두 소화하기 위하여 앞에서도 언급한 바와 같이 하루 네 번씩 음악회가 열리는 그야말로 강행군의 일정이었다. 또한, 9월 말임에도 불구하고 한낮 기온이 삼십 도를 오르내리는 북텍사스대학 캠퍼스를 하루에 수 킬로미터씩 도보로 이동하며 음악회를 참석하는 일도 쉽지만은 않았다. 실제로 처음 며칠은 시차와 피곤함이 겹쳐서 음악회 중간에 밀려오는 졸음을 참느라 연주되는 작품을 제대로 감상할 수 없었던 경우도 종종 있었다. 하지만 이번 음악제를 통하여 미국 및 유럽 그리고 아시아 각국의 신진작곡가와 기성작곡가, 그리고 학생까지 포함한 다양한 세대와 다양한 문화권에 기반한 음악을 접할 기회를 가지게 되어 소중한 경험을 했다.

학회에서 열린 모든 음악회들을 빠짐없이 참석하는 것은 실제로 많은 어려움이 따랐기 때문에 모든 작품들을 전부 들어볼 수는 없었지만, 필자가 감상했던 작품들 중 많은 수가 훌륭한 아이디어와 완성도 높은 작품성을 가진 것들이었다. 그 중에서 몇몇 기억하는 작품들의 예를 들어보면, 케이트 키르코프Keith Kirchoff의 《내일의 프리즘을 통해 과거 보기Seeing the Past Through the Prism of Tomorrow》, 조지-그레고리오 가르샤 몬카다Jorge Gregorio Garcia Moncada의 《우리의 역사La historia de nosotros》, 코디 카울Cody Kauhl의 《여담: 세 예술가곡Excursus: Three Art Songs》, 존 깁슨John Gibson의 《붉은 깃털Red Plumes》, 존티 해리슨Jonty Harrison의 《부적절한 대상Unsound Objects》, 제이슨 볼테Jason Bolte의 《눈을 감고With My Eyes Shut》, 폴 보텔로Paul Botelho와 존 애플톤Jon Appleton의 공동작품인 《네어수레리N'air sur le lit》, 마라 헬무스Mara Helmuth의 노트북 앙상블 작품인 《우간다에서from Uganda》, 세스 샤퍼Seth Shafer의 《펄서 [변주II]Pulsar [Variant II] 》, 파나요티스 코코라스Panayiotis Kokoras의 《티-토텀T-Totum》 그리고 래리 오스틴Larry Austin의 《리덕스투ReduxTwo》 등이다. 이 작품들 외에도, 여러 측면에서 강한 인상을 심어주기에 충분한 곡들이 상당수 있었다.

지난 1981년 북텍사스주립대학에서 열린 제7회 학회의 의장이었던 오스틴은 여든 다섯의 고령임에도 불구하고 음악제 참가자로서 거의 모든 연주회에 참석하는 등 일정을 빠짐없이 소화하였다. 마지막 음악회에서는 자신의 작품을 발표하면서 컴퓨터 음악에 대한 노익장의 열정을 유감없이 보여주어 주위 사람들로부터 기립박수를 받기도 하였다. 이러한 그의 열정이 필자에게 무척이나 깊은 인상이 남아 지금도 뇌리에 선명히 떠오른다. 여드레의 짧지 않은 기간 동안 컴퓨터 음악과 관련된 많은 사람들을 이 학회를 통해 만났고, 몇몇과는 적지 않은 이야기를 나누었다. 필자가 유학시절 같이 공부했던 친구들과 은사님과도 오랜만에 만나 반가운 인사를 나누었고, 이들과 음악제가 끝나는 시각까지 유익하게 시간을 보낸 것이 이번 학회에서 얻은 수확 중의 또 다른 점이라 하겠다. 세계 각지에서 모인 각양각색의 사람들과 컴퓨터 음악이라는 하나의 공통점을 두고 며칠 동안 같은 장소에서 함께 시간을 보낸 것은 색다르고 재미난 경험이었다.

A Review of *Linux Audio Conference 2015*

Kim, Jonghyun

The *Linux Audio Conference (LAC)* is an international conference aimed at the developers, composers, artists, and audio technicians of Free Open-Source and GNU/Linux-based Software. GNU/Linux is an open, stable, and professional platform for audio and media researches and musical production. The conference has been held annually since 2003, revolving usually around Europe and U.S. This year, the 13th meetings of *LAC* took place in the *Johannes Gutenberg University (JGU)* at Mainz, Germany for four days of April 9-12, sponsored by the *IKM (Institut für Kunstgeschichte und Musikwissenschaft)* of the *JGU* and organized by Albert Gräf and his team in the *Computer Music Research Group (Bereich Musikinformatik)* of the *IKM*. One of the noticeable paper presentation was on FAUST(Functional AUdio STream), a functional audio programming language developed by *Grame* and *CCRMA* since 2002. The concert works of *LAC* characterized as composed exclusively through open-source software and programming languages such as Pure-Data, SuperCollider, Csound, and Ardour. The author, as a second consecutive year participant, presented a composition for motion sensors and wii-remotes with the treatment of granular synthesis. I would like to recommend joining *LAC*, especially for such musicians as pleased with open-source in the process of creating their music.

리눅스오디오학회2015 참관기

김종현

1. 학회 개요

리눅스오디오학회*Linux Audio Conference(LAC)*는 오픈소스 소프트웨어와 리눅스오에스OS(GNU/Linux)를 기반으로 한 음악소프트웨어의 개발자, 작곡가, 예술가, 음향 엔지니어를 대상으로 한 국제학회이다. 필자에게는 2014년에 이어 두 번째 참여인데, 올해에는 위리모트Wii-Remote와 그래뉼러합성Granular-synthesis을 사용한 모션센서 작품을 발표하였다. 이 학회는 2003년부터 매년 유럽과 미국을 중심으로 개최되어 왔으며 .열 세번 째인 올해는 4월 9일부터 12일까지 나흘 동안 독일 마인츠 구텐베르크대학*Johannes Gutenberg University (JGU) in Mainz*에서 개최되었다.

1477년에 설립된 마인츠 구텐베르크대학은 독일에서 가장 오래된 대학 중 하나로 규모는 독일 내 열 손가락 안에 든다. 독일의 국립대학으로는 특이하게 음악대학Musikhochschule과 미술대학Kunsthochschule을 포함하고 있다. 구텐베르크대학이 올해 학회의 주최자로서 동 대학의 음악정보학연구소*Musikinformatik* 컴퓨터음악연구팀*Computer Music Research Group*이 조직위를 맡았다.

위에 언급한 음악정보학연구소로 칭한 단어 'Musikinformatik'은 최근 독일과 미국의 대학에서 시도하고 있는 융합학문으로 음악과 컴퓨터 과학이 합쳐진 분야이다. 한국어 번역명은 아직 없지만 '음악정보학' 정도로 번역될 수 있는데, 음악테크놀로지Music Technology(MT) 분야와 혼동되기 쉽다. 컴퓨터음악연구팀은 1991년에 설립되었으며 독일 최초 음악과 컴퓨터 과학의 융합 연구소로, 오픈소스와 리눅스를 음악 분야에 활용하는 연구가 활발한 곳으로 알려져 있다.

처음 학회의 이름을 들은 사람들은 리눅스 사용자나 개발자 만을 대상으로 할 것이라 착각할 수 있다. 학회 이름에 붙은 '리눅스'는 오픈소스 소프트웨어 운동에서 가장 중요하고 상징적인 프로젝트, '리눅스오에스'에서 따온 것이며, 반드시 리눅스

사용자가 아니어도 오픈소스 관련된 프로젝트라면 모두 포괄한다. 리눅스라는 이름은 상징적인 의미로 오픈소스와 관련된 모든 음악 소프트웨어 분야에 사용된다. 모든 참가자들이 리눅스만 사용하는 것은 아니며, 맥오에스MacOSX 와 윈도우Windows 사용자도 많다. 대부분의 오픈소스 소프트웨어는 리눅스Linux, 맥오에스MaxOS, 윈도우Windows, 안드로이드Android, 아이오에스iOS를 가리지 않는 플랫폼 독립적 모델Platform-independent model을 따르고 있기에 모든 운영체제의 사용자가 혼재되어 있다. 하지만 리눅스가 제외된 맥, 윈도우 전용의 상업용 음악 소프트웨어의 개발과 사용은 해당 주제가 아니다.

학회의 구성은 국제컴퓨터음악학회*ICMC*와 같이 일반적인 컴퓨터 음악 학회와 큰 차이점은 없다. 구성은 페이퍼 발표, 포스터 세션, 워크샵, 음악회, 사운드인스톨레이션, 라이브코딩세션으로 이루어져 있다. 타 컴퓨터 음악 학회와의 차이점은 학회 특성상 프로그래머가 많기때문에 그들에게 선호되는 라이브 코딩 음악Live Coding Music 분야가 따로 있다는 것 정도이다. 참가자들의 국적 분포는 대략 유럽 70%, 미국 25%, 기타 5% 로, 올해는 독일에서 개최된 이유로 유럽인들이 많았다. 동아시아(한국, 중국, 일본)를 통틀어 참가자는 필자 뿐이었는데, 이는 오픈소스 프로젝트가 대부분 유럽과 미국인들이 주도하고 있으며 아직은 동아시아에서의 인지도가 저조한 것을 보여준다.

2. 논문주요발표

리눅스오디오학회2015에서 눈에 띄는 논문발표는 파우스트(기능성 오디오 스트림)FAUST(Functional AUdio STream)에 대한 내용이었다. 파우스트는 프랑스의 음향 연구소인 그람*grame*과 미국의 스탠포드 칼마 *CCRMA(Julius O. Smith)*를 주축으로 2002년부터 오픈소스로 개발하고 있는 함수형 오디오 프로그래밍 언어functional audio programming language이다. 이것은 시플러스플러스C++로 작성되어 성능이 뛰어나고 태생 자체가 플랫폼 독립성을 목적으로 제작되어 리눅스, 맥오에스, 윈도우, 안드로이드, 아이오에스, 웹 등 현존하는 거의 대부분의 운영체제를 지원하기때문에 한번 코드를 작성하면 모든 플랫폼에서 작동하는 것이 특징이다.

파우스트 관련 논문발표는 총 세 번에 걸쳐 이루어졌는데, 첫 번째는 웹페이지 상에서 구현하는 파우스트 웹 오디오 에이피아이FAUST Web Audio API 사용 방법, 자바스크립트JavaScript로 구현하는 방법 등을 소개하였다. 두 번째는 아이오에스와 안드로이드 스마트폰 어플리케이션 제작에 관한 내용으로 실시간 오디오 처리에 있어 저지연성low-latency 구현, 멀티터치 사용자 인터페이스, 가속도계Accelerometer사용법, 오에스시/미디OSC/MIDI통신 방법론을 설명하였다. 세 번째로 파우스트 디에스피FAUST DSP 언어의 본질에 관한 수학적 이론과 함수형 언어의 특징을 설명하였다.

논문발표는 별개로, 리눅스오디오학회에서 보기 드물게 상업용 소프트웨어인 비트윅 스튜디오 디에이더블류Bitwig Studio DAW 워크샵이 있었다. 비트윅 스튜디오 디에이더블류는 오픈소스 큐티 프레임워크QT framework를 사용하여 리눅스, 맥, 윈도우에서 동일한 지유아이GUI를 제공하는 것이 특징이다. 리눅스를 지원하는 상업용 소프트웨어는 특히 음악분야에서 극히 드물기에 주목을 받았다.

3. 음악회 주요발표

리눅스오디오학회 음악회의 특징은 오픈소스 소프트웨어 및 퓨어데이터Pure Data, 슈퍼콜라이더SuperCollider, 시사운드Csound, 아더Ardour 등의 프로그래밍 언어로 만들어진 작품만을 다룬다는 것이다. 라이브로 연주되는 작품과는 다르게, 고정매체Fixed Media 작품은 오디오파일이나 미디어만 제출하기 때문에 사실상 오픈소스 소프트웨어만 사용했는지 알 수 없거나 의심이 가는 작품도 있었지만, 작곡가의 설명을 믿는 수밖에 없다고 생각했다. 전체적인 작품의 품질은 국제컴퓨터음악학회나 기타 유명학회에 뒤지지 않을 만큼 동등한 수준이라고 평가된다. 대부분의 음악회는 구텐베르크대학(종합대학) 내 마인츠음악대학*Mainz Hochschule für Musik* 콘서트홀에서 진행되었다. 친목파티를 겸한 레이트나잇Late Night 프로그램은 콘서트홀이 아닌 퍼브Pub에서 진행되었고, 사운드인스톨레이션은 대학 내 여러군데에 분포되어 전시되었다.

마인츠음악대학은 일반적인 독일의 음대와 달리 종합대학 내에 있으며, 기악음악 작곡과Komposition가 없고, 대신 사운드아트작곡 *Klangkunst-Komposition* 학과가 있다. 이 학과는 2009년에 신설되었고 이 년제 석사학위M.Mus 과정이며 현대음악, 전자음악, 사운드아트, 오디오비주얼 등 융합 예술을 추구하는 것으로 보인다. 독일 대학 답지 않게 입학시험 시 독일어능력이

아닌 영어능력증명*Englische Sprachkenntnisse*을 요구한다.

아쉽게도 사운드인스톨레이션 작품들을 모두 다 보지 못했다. 대학 곳곳에 설치되어 있었는데 구텐베르크대학의 면적이 너무 넓어서 장소를 찾기 어려웠고 필자 자신의 작품 발표 준비에 걸어서 멀리 이동할 겨를이 없었던 것이 이유였다. 아쉬웠지만 다음을 기약한다.

4. 참관소감

구텐베르크대학 주최의 2015년 학회는 예술미디어테크놀로지센터*Zentrum für Kunst und Medientechnologie(ZKM)*에서 치뤄진 전년도에 비해 조금 아쉬웠다. 구텐베르크대학의 기획과 배정 장소가 그리 나쁘지 않았으나 예술미디어테크놀로지센터 *ZKM*가 독일 최고의 음향 멀티미디어 연구소로서 최 정상급 연주홀과 멀티미디어 시스템을 갖추고 있었기에 그와 비교되는 것은 어쩔 수 없다고 생각한다.

차기 리눅스오디오학회 2016은 독일 베를린에서 개최된다. 오픈소스를 사용하여 작품을 만드는 음악가라면 내년이 아니라도 한 번쯤은 꼭 참가해 보길 추천한다. 여느 학회와 달리 논문게재료와 심사료가 없고 참가비도 받지 않는다. 오픈소스를 사랑하는 사람이라면 누구에게나 열려 있는 곳이다.

Young Frontier of New Sound:
Review of *ARKO Young Artist Frontier 2015*

Park, Soon-young

Performing arts can make audience to feel a great change in art and technology as time passed, which would come, however, not a moment unless representing energetically the outcome as condensate and accumulation within the time limit. *ARKO Young Artist Frontier (AYAF)* supported by *Arts Council Korea* is a performance project to lay out the ground for young artists who would acknowledge the nature of artistic performance, create their works continuously, and find the way to the future new music. They are sponsored for a year with research investigations, overses trips, performing and composing in concerts, mentoring programs, etc. Some of the twenty artists of *AYAF 2014* were Jinok Cho, Sangbong, Seung-yeon Nam, Tae-won Kim, and Dong-gyu Lee, all of whom impassionedly worked for the area of sound and electronic music. The ways of dealing with music and sound varied from using cutting-edge interactive technologies to attempting unconvertionally conceptual approaches to simple ideas resulting in new sound.

새로운 소리의 젊은 개척자들:
아르코 차세대 예술인력 육성사업 공연 2015 참관기

박순영

문화 예술의 흐름과 변화는 어느 날 어느 시점의 한순간에 이루어지는 것이 당연히 아니다. 하지만, 공연예술은 지난 기간 동안 축적되었던 에너지를 제한된 공연 시간 안에 응축해서 표현해내야 하는 장르의 특성상, 세월의 흐름을 통한 문화 예술과 기술의 변화를 고스란히 한 작품 안에서 느끼도록 해야 한다. 한국문화예술위원회 차세대예술인력 육성사업(아야프)ARKO Young Artist Frontier(AYAF) 공연예술 창작자 부문은 이러한 공연예술의 특성을 잘 살려, 앞으로 우리 문화 예술의 미래를 선도할 만35세 이하 젊은 예술가들에게 약 한 해 동안의 연구조사, 해외리서치, 공연 창작, 멘토링 등의 지원을 통해 지속적인 창작작업에 토대를 마련해 주는 사업이다. 지난 2010년 시작되어 2014년까지 다섯 번 시행되었으며, 2013년부터는 기존의 창작지원에 멘토링을 더해 신진 예술가들의 유대와 필요를 더욱 헤아린 사업을 진행 중이다. 2014년에는 무용, 연극, 전통, 음악, 다원의 다섯 개 분야에 백스무 명, 2015년에는 보다 많은 백예순아홉 명이 지원했으며 두 해 모두 스무명의 신진예술가가 선발되었다.

지난 해 아야프2014의 한 해 사업의 결과로 예술가들의 공연 및 발표가 2014년 12월 20일부터 2015년 2월 1일까지 진행되었다. 아야프 2014 스무 명의 예술가 중 사운드, 전자음악과 관련된 이는 음악부문의 남상봉, 다원부문의 조진옥, 남승연, 김태원, 이동규, 총 다섯 명이다. 첨단 테크놀로지를 통한 인터랙션부터 간단한 기술이지만 기존관념의 탈피라는 개념적 접근을 통한 새로운 시도까지, 각 작가와 작곡가별로 음악과 소리를 다루는 방식은 다양했다. 소재별로는 한국 전통 소재 두 편, 일상 관련 두 편, 밤에 대한 주제가 한 편이고, 유형별로는 음악회 형태의 한 작품, 멀티미디어극 세 작품, 전시형태 한 작품으로 분류할 수 있다.

남상봉『밤: 인시』, 엠포이mPoi가 만든 밤의 음악극

1월 28일 문화역서울284 RTO공연장에서 작곡가 남상봉의『밤: 인시』공연이 열렸다. 이 공연은 바이올린, 첼로 등의 악기와 컴퓨터, 조명, 그리고 그가 유학시절부터 다섯 해에 걸쳐 직접 개발한 악기인 엠포이 등 각각의 독특한 개성이 모여 하나의 음악극을 이루는 수준 높은 공연이었다. 소셜네트워크서비스SNS에서 홍보된 엠포이 개발과정에서부터 공연 삼십 분 전 이미 홀을 꽉 메운 관객수까지 이에 대한 특별한 관심이 느껴졌으며, 패션쇼장처럼 공간의 가운데를 세로로 길게 무대로 하고 양 옆의 객석이 서로 마주보게 하는 새로운 배치형태가 공연을 더욱 기대하게 만들었다.

한 시간의 공연은 '밤'의 고요와 고독에 대한 단상을 기 드 모파상Guy de Maupassant의 문학작품《그날밤La Nuit》을 중심으로, 작곡가의 이전 음악 작품들과 엠포이 퍼포먼스로 구성된 아홉 개 부분의 음악극으로 풀어냈다. 조명이 어두워지고 첫 곡《나잇-캄Night-Calm》이 무대 뒷 편 높은 단상 위에서 작곡가 신지수의 토이피아노로 시작되었다. 고요함을 깨트리는 저주파 전자음향의 진동 속에 토이피아노 금속성의 맑은 음색과 아르페지오 움직임이 신비로웠다. 천장에서 공연장 전체를 비추며 천천히 회전하는 은색 조명으로 마치 관객들은 별빛 우주 속에 앉아있는 느낌이다.《엠포이솔로mPoi Solo》는 움직일 때마다 빨강 파랑 초록의 갖가지 색을 띄는 엘이디LED빛도 멋지지만, 어두컴컴한 가운데 오직 엠포이의 불빛만이 그려내는 다양한 원형의 궤적이 실로 놀라웠다. 남상봉이 포이 아티스트Poi Artist이자 파이어댄서fierdancer, 정신엽과의 긴밀한 협업으로 찾아낸 엠포이의 변화무쌍한 운동성과 소리의 질감으로 이루어진 싱크로니제이션이 실시간 퍼포먼스를 통해 그 탁월성을 입증하였다. 맥스/엠에스피 Max/MSP, 그래픽 프로그래밍 언어Processing, 가속도Acceleration 및 자이로Jiroscope 센서, 엑스-오에스시x-OSC 등을 활용한 엠포이의 움직임은 원형, 누운 팔자형, 에스S자형, 에스자형의 연속 등 실로 무한했다. 원운동이 천천히 아래쪽에서 시작할 때는 저음의 엔진 시동음 같은, 빠르게 원운동을 지속할 때에는 일렉트릭 기타의 최고음역대를 빠른 패시지로 연주하는 듯한, 바람소리와 레이저광선, 나무조각, 돌조각 등의 여러 질감의 소리를 오실레이터로 만들어 다양한 음색으로 표현했으며, 다양한 운동의 모습을 발광다이오드LED빛으로 나타냄으로써 청중의 눈과 귀를 사로잡았다.《어웨이큰Awaken》은 공연의 제목『밤: 인시』의 '인시(새벽 3시-5시)'에 절의 잠들어있는 세상 속 사물들을 깨우기 위해 사중사물(범종, 법고, 목어, 운판)을 두드리는 것에서 영감을 얻어 씌여진 곡이다. 절에서 사용하는 주발 소리를 녹음한 음원이 전자음향으로 전환되어 몽롱하고 고요하게 울리는 가운데, 작가 올리버 그림이 고안한 무대 천장의 거울이 내는 작은 반사빛이 지속적으로 무대 앞뒤를 느리게 회전한다. 절에서 백팔배를 반복해 드리듯 어둠 속 간절한 염원과 맑은 정신이 느껴지는 곡이었다. 이어서 무대 뒤 통로에서 시작된 바이올린 독주곡《보이지않는 움직임Invisible Movement》은 빠르고 느린 패시지의 대조, 격렬하거나 조용하면서도 빠른 부분의 대조가 돋보였다. 복잡한 음계의 패시지나 선율의 구조보다는 바이올린 현 사이를 이동하거나 크게 연주된 음형이 곧바로 이어 작게 반복하는 등 오른손의 활과 왼손 손가락의 움직임이 두드러짐을 느꼈다. 공연의 전체 구성에서 유일하게 전자음향이 없는 독주 기악곡으로 신선함을 더했다.《시프트 2번Shift No.2》는 컴퓨터가 기존 음악을 변화, 확장시키는 의의와 함께, 컴퓨터의 시프트 키처럼 더블베이스 특유의 메마르고 묵직한 저음역 보잉이 컴퓨터를 거쳐 전자음향으로 변환되면서 스피커 사이를 회전했다. 이어 토이피아노의《나잇-론리Night-Lonely》가 다시 무대 뒤 원형 조명으로부터 들려왔다. 토이피아노의 점묘적인 음형이 딜레이와 리버브 효과로 변조되어 고독감을 드러냈다.

《엠포이 앙상블mPoi Ensemble》은 공연 앞 부분의《엠포이솔로》의 격렬함과는 다르게 차분함과 고요함을 보여주었다. 엠포이 원운동의 주기성과 무한성에서 '패턴의 반복'이라는 착상을 가지고 명상적인 소리와 결부되었다.《엠포이솔로》가 보인 원운동과 소리 사이의 상호작용이 없는 대신, 엠포이 세 대가 동시에 그리는 원운동의 다양한 모습, 잔잔한 전자음향의 진동과 더불어 다채로운 빛깔의 조화로운 모습에서 '무아'의 상태가 떠오르기도 했다. 다음으로《나잇-너버스Night-Nervous》는 밤의 고독을 벗어나기를 원하는 이면의 두려움을 바이올린과 첼로의 하모닉스 등의 현대주법과 전자음향으로 표현했다. 마지막 순서의《언베어러블Unbearable》은 바이올린, 더블베이스, 플루트, 클라리넷, 타악기, 세 대의 엠포이, 그리고 전자음향이 함께 어우러진 그야말로 이색적인 장면이었다. 어슴프레한 고요가 끝나가고 동트는 새벽의 기운이 타악기 우드블럭의 미묘한 움직임, 현악기의 하모닉스와 하행음계, 목관악기의 지속음으로 대체되었다. 점차 동음 반복 트레몰로가 악기 간 번갈아 나타나고 격렬한 전자음향의 회오리도 함께 휘몰아치더니 타악기의 격렬

한 비트와 현악기의 글리산도 등이 포르티시모로 지속되었다. 악기연주와 전자음향이 최고조에 다다랐을 때 세 명의 엠포이 앙상블의 현란한 불빛의 움직임도 함께 등장해 듣는 것과 보는 것의 혼연일체가 밤에 대한 황홀감을 안겨주며 대단원의 막을 장식했다. 공연의 마지막 순간 대담하고 거침없는 조합이 꽤 인상적이어서 한동안 더 지속되었으면 좋겠다는 생각도 들었다.

아야프의 체계적인 지원 덕에 지난 일곱 달 간 공연을 준비해 올릴 수 있었다는 남상봉은 "음악극의 특색을 유지하면서 나의 이전 작품들을 독립적으로 선보이는 것에 초점을 두었다"면서 "전자음향의 극대화보다는 악기와 전자음향, 그리고 엠포이 솔로와 엠포이 앙상블 각 요소간 균형에 역점을 두고 준비했다"고 공연취지를 설명했다. 지난 다섯 해의 유학시절 대표적 작품인 해외 작곡 콩쿨 수상작과 개발악기를 위한 작품을 한 자리에 모아 종합선물세트 음악극이자 자신만의 리사이틀로서 훌륭하게 연출하는 능력이 돋보였다. 작곡 리사이틀과 음악극의 경계를 사뿐히 넘으면서 자신의 의도와 면면을 욕심껏 펼쳐내어 보일 수 있는, 배짱이 두둑한 그의 다음 공연 역시 기대된다.

조진옥 멀티미디어 음악극《수궁가》, 어린이도 볼 수 있는 24개 스피커 수궁가

1월 24일 경기도 일산 아르코예술인력개발원 실험무대에서 열린 조진옥의 멀티미디어 음악극 《수궁가》는 판소리가 영상 및 입체음향과 결합한 한 시간 여의 공연이었다. 어려운 옛말의 판소리를 오늘날의 쉬운 우리말로 풀고, 각 대목에 맞는 영상으로 좀 더 친숙하고 재미있게 판소리를 이해할 수 있었다.

본 공연에 앞서 스코트 와이어트Scott A. Wyatt, 김동선, 조진옥, 고병량, 송향숙, 다섯 명 작곡가가 24채널 라우드스피커 오케스트라를 위한 창작곡을 발표 및 해설하는 렉처 콘서트가 열렸다. 상업 영화관이나 홈씨어터 시스템을 통해 접할 수 있는 5.1채널 서라운드 스피커 시스템은 익숙하지만, 스물 네 개 채널의 스피커가 입체적으로 청중을 둘러싸고 있는 장치에 관객들은 상당한 호기심을 드러냈다. 각 작곡가의 작품마다 스피커를 따라 소리가 순차적으로 한 방향으로 이동하기도 하고, 여기 저기 곳곳에서 발생하기도 하는 등 음원의 위치와 그 이동 방향, 깊이감 등을 다채롭게 느낄 수 있다는 점에 관객들은 큰 호응을 보였다.

이어 본공연에서 관중들이 더욱 늘어났다. 판소리 수궁가 완창을 들어본 사람이라면 아마도 그들은 국악 전공자 중에서도 판소리 전공이거나, 대단한 국악 애호가, 아니면 꽤 연배가 높은 어르신일 것이다. 그 만큼 요즈음은 연주 횟수가 적기 때문에 우리의 문화유산을 전통 그대로의 형태를 접하는 것이 힘들고, 원형 그대로라고 하더라도 스마트폰, 컴퓨터, TV 등 다양한 전자기기와 멀티미디어 방식에 익숙한 사람들에게 옛 방식의 예술문화를 소화하는 것은 어려움이 있다. 하지만, 어떤 어린이가 이 날 공연의 《수궁가》를 보았다면 "아, 토끼랑 거북이 그림이 나오네? 수궁가 재밌다!"라고 했을 것 같다. 세 시간 반의 완창본 수궁가를 한 시간으로 줄여 가사를 정리하고, 용왕, 바닷 속, 토끼, 거북이 세계를 위한 대금선율을 각각 만들고, 용왕이 타고 오는 구름을 전자음향과 함께 수묵담채화로 은은하게 그렸다. 주요대목에 등장하는 영상(영상디자인: 최성민)은 현란하거나 빠르지 않고 천천히 화면을 이동해, 관객에게 극의 흐름을 이해하는데 도움을 주되 음악의 흐름에 방해되지 않도록 했다. 파리의 음악연구팀Groupe de Recherches Musicales(GRM)의 '아쿠스모니엄Acousmonium', 영국 버밍엄대학의 '비스트BEAST' 등 해외 유명 음향기관에서는 다채널 스피커시스템의 연구가 활발하지만, 우리나라에서 이를 연구한 후 대중을 위한 공연에 연결한 적은 이번 조진옥의 시도가 처음인 듯 하다. 이번 공연 사운드 시스템의 조절하기 위해서 서울대학교의 예술과학센터 책임연구원 고병량이 개발한 프로그램 '아쿠스블렌더AcousBlender'가 사용되었다. 더불어, 정은혜의 소리와 김인수의 북, 이아람의 대금은 찰떡같은 노래와 연주로 몰입감을 주었다. 일반적인 2채널-스테레오 스피커가 아닌 앞, 뒤, 옆, 위 사방팔방의 24채널 라우드스피커 오케스트라로 실연된 소리는, 용왕이 구름을 타고 내려올 때는 천장스피커에서 아래쪽으로, 거북이 토끼를 잡으러 바다에서 육지로 올라갈 때는 거대한 소리물결이 공연장 스피커 뒤쪽에서 앞쪽으로 서서히 이동하고, 토끼와 함께 바다로 돌아올 때는 바다물살이 앞에서 뒤로 첨벙거린다. 이러한 소리와 현장감을 더하는 영상으로 관객은 바다 속 용궁세계를 거북의 등을 타고 경험한다. 극의 전체적 균형을 위해 기지로 위기를 모면한 토끼의 모습을 노래한 "토끼화상" 대목을 맨 마지막에 다시 배치한 것이 인상적이었다.

조진옥은 "오늘날 '시각적' 자극에 매몰되어 있는 우리 어린이들이 '듣는 것'에 대한 재미를 찾고 쉽게 이해할 수 있도록, 판소리의 보편적 이해를 높이는데 노력했다"면서, "제가 작곡가로서 돋보이는 것보다는 저의 음향적 기술이 판소리 수궁가를 이해하는 데 조금이라도 도움이 되었으면 하는 심정으로, 한마디로 '양념'을 친 거죠"라며 작품 의도를 밝혔다. 1990년대 초 한 TV광고에서 생전의 박동진 명창이 "우리 것은 소중한 것이여"라고 하던 대목이 생각난다. 그로부터 스무 해쯤 지난 지금 '우리 것'은 현재와 결합해 더욱 다양한 모습으로 변모했다. '우리 것'의 원형과 '현대적인 것'을 어떤 형태와 비율로 결합하여 무엇을 만들지가 중요한 데, 작곡가 조진옥의 멀티미디어 음악극《수궁가》 속 '그가 친 양념'은 전통의 수궁가를 다시 한 번 바라보고 이해하는데 모자라지도 넘치지도 않게, '알맞게' 더해진 것이었다. 요리의 전체 과정에서 제일 중요한 사람은 마지막에 간 맞추는 사람 아니던가? 그가 다음에는 어떤 음식을 선보일지 기대된다.

이동규《원이 엄마》, 우리정서와 몸짓이 만난 멀티미디어극

이동규《원이 엄마》는 1월 17일 아르코예술인력개발원 실험무대에서 상연되었다. 이동규는 '원이 엄마'라는 전통이야기의 소재를 한국무용과 사운드, 인터랙티브 영상, 조명의 첨단 테크놀로지를 활용해 공감각화시켜 열 개의 장scene, 한 시간의 공연으로 탄생시켰다. '원이 엄마' 이야기는 사백이십 여 년 전 안동지역에서 전해내려오는 실화로, 1998년 그 지역의 택지개발지구에서 무덤 이장 중 이응태의 무덤이 그의 아내인 '원이 엄마'가 쓴 편지 한 장과 함께 발견되었는데, 이 내용에 근거한다. 공연장에 신비롭고 어두운 분위기가 감도는 가운데 관객 입장에서부터 이미 공연은 시작되었다. 화려한 의상의 꼭두(오성근)가 입장하는 관객들을 향해 마치 영혼이라도 빨아들이듯 팔을 휘두르며 공연석으로 인도하고, 키넥트Kinect(3차원 모션캡쳐 카메라)가 관객의 움직임을 실시간으로 트래킹Tracking하여 무대 검은 장막 위로 흰색 외곽선이 투사한다. 피아노 상성의 움직임과 북의 강렬한 고동소리, 조명을 한껏 받은 꼭두의 카리스마 넘치는 춤사위가 무대 위로 올려진다. 오른편으로 움직이던 꼭두가 음악에 맞추어 주술사처럼 팔과 손을 휘저으며 의미심장하게 모래를 파헤친다. 실시간 카메라 기법으로 모래그림과 움직이는 손의 모습이 무대 화면 가득 보였는데 관객이 마치 무덤 속에서 무덤이 파헤쳐지는 것을 올려다보는 것 같은 느낌을 주었고, 이는 앞으로 전개될 뒤엉킨 인생과 사랑의 모습을 암시하는 것이었다. 곧, 징소리가 들리며 푸른 조명 아래 하얀 소복을 입은 원이 엄마(남미경)가 등장한다. 묵직한 피리의 처연한 음악은 감각적이면서도 전통의 향취가 묻어난다. 키넥트가 감지한 무용수의 동작이 맥스포라이브Max for Live를 통해 음악의 고음역대 음색을 미세하게 변조(사운드 인터랙션 김준환)시킨다. 이에 맞추어 원이 엄마의 살풀스러운 춤은 감각적이고 애틋하다. <기억-시간> 장면이 텍스트와 함께 열리며 겹쳐지는 원이 엄마와 이응태(윤한엽)의 듀엣이 무척 아름답다. 원이 엄마의 편지에서처럼 세상에 하나밖에 없을 아름다운 사랑이 느껴지는 춤이다. 배경에는 파티클(컴퓨터그래픽 폴리곤의 일정 부분)이 흩어졌다 모아지며 선을 이루는 추상적인 영상이 무용 동작과 잘 어울린다. 파티클의 적산(積散) 반복되는 형상은 키넥트가 인식한 무용수의 팔 동작을 그래픽 프로그래밍 언어 Processing로 만들었는데(영상인터랙션: 이재중), 무용수가 손에 든 부채를 펼치며 팔을 더 넓게 휘두르면 파티클이 그만큼 많이 흩날리는 결과를 낳는다. 어두운 저음의 사운드가 들리며 이응태가 쓰러진다. 극의 처음부터 줄곧 장막 뒤편에 꼭두가 두려움에 떨며 모래를 파헤치며 그림자처럼 드리운 모습이 계속되는데, 음산한 소리를 배경으로 원이 엄마의 독무, 이응태와의 복무 등으로 이루어진 <경계> 장면에서는 원이 엄마의 자필편지가 영상으로 희미하게 드러난다. 시계의 재깍거리는 소리가 들리고 꼭두는 원이 엄마를 감싸며 어두운 기운을 불어넣는다. <엮음> 장면이 시작된다. 구슬픈 대금소리와 함께 꼭두는 붉은 천과 길다란 붉은색, 노란색, 파란색, 녹색의 띠를 하나씩 가지고 나와 원이 엄마에게 묶는다. 뒤편은 띠 모양 영상이 비춰지고, 원이 엄마는 갖가지 색의 띠를 엮어 한데 쥐고, 이응태에 대한 온 사랑을 담아 춤을 춘다. 장사 지내는 모습의 영상이 검은 배경에 흰색 외곽선으로 추출되어 보이고, "웅"하는 진동음을 바탕으로 상여소리가 들린다.

<혼> 장면에서는 병상에 쓰러진 이응태가 비치는 검은 장막 뒤에 누워있다. 애달픈 대금소리와, 징, 북, 전자음향의 옅은 고동과 함께 꼭두가 심장을 죄여오듯이 이응태를 무섭게 휩싸며 마술사처럼 주문을 건다. 혼이 빠져나가는 것처럼 이응태의 모습이 누워있던 몸 위로 올라가 잠시 머물다 사라지는 것이 검은 장막에 카메라 모션 트래킹 기법으로

선명히 보여진다. 몸을 떠난 이응태의 영혼이 푸른 횃불의 모양으로 장막 이곳 저곳을 휘돌아 다니는데, 이것은 장막 뒤에서 춤추고 있는 꼭두의 팔 동작 위치 값을 키넥트로 인식해 푸른 빛과 실시간 동기화 한 것이다. 꼭두에게 이끌려 이응태는 무대 밖으로 홀린 듯 걸어나가며 죽음에 이른다. 잔잔한 피아노 음과 푸른 조명 빛을 배경으로, <윤회> 장면에서는 흰색소복의 원이 엄마가 자신의 머리카락으로 소중히 엮은 미투리를 손에 쥐고 등장해 이응태에게 신겨준다. 앞의 복무에서 보였던 영상의 파티클이 보이다가 청사초롱으로 변한다. 펄럭이는 장막 뒤에서 둘의 마지막 춤이 너울거리며 만나고 떠나기를 반복하는 모습으로 분위기를 고조시킨다. 영상 속 꽃가루가 날리면서 사백이십 여 년 전 원이 엄마가 실제로 썼고 이응태와 함께 무덤에 묻혀 있던 자필편지의 영상이 보여지고 편지내용 자막이 올라가며 공연은 끝난다. 이동규는 "제가 안동 출신인데, '태양의 서커스'처럼 대중적이면서 자신들의 이야기를 몸동작으로 풀어낸 멋진 퍼포먼스를 하고 싶었어요"라며, "우리 주변의 이야기를 우리의 동작이나 사운드로 풀어내면 확실히 다르지 않을까 생각해서 안동의 이야기인 '원이 엄마'를 택했다. 여기에 한국무용과 동작에 반응하는 소리와 영상이 결합된 인터랙티브 멀티미디어 공연으로 기획했다"라고 설명했다. 시적 감성이 느껴지는 한국무용, 영상과 음악에 동작이 연동되는 테크놀로지의 결합, 그 곳에는 사운드와 음악을 중심으로 바라보고 현대적 감각으로 전통소재와 멀티미디어를 결합해 낸 이동규가 있었다.

남승연《장면(場面)》, 창과 깊이를 통한 일상의 소리 재발견

남승연은《장면(場面)》이라는 제목으로 사십 분의 사운드 퍼포먼스를 준비해 1월 29일부터 1월 31일까지 아르코예술인력개발원 실험무대에서 상연했다. 이 공연에서는 아르코예술인력개발원 실험무대 공연장 전체를 사용하지 않고, 삼사십 여 명의 관객이 들어갈 수 있는 크기에 흰 색 천으로 사면을 막은 커다란 정육면체 구조물을 건물 안에 만들었다. 그 둘레로 의자들이 놓였고, 그 가운데 테이블 한 개가 놓인 곳에서 공연이 진행됐다. 또 한 가지 이색적인 것은, 스피커가 아니라 헤드셋으로 소리를 듣는 것이었는데, 청중은 다른 관객들과 함께 같은 연극적 공간 안에 존재하지만, '헤드셋'이라는 매개체를 통해 온전히 분리되는 개별적 체험을 하게 된다. 남승연이 대표를 맡고 있는 프로젝트《남김》은 일반 공연장 외에 길거리를 이동하며 헤드셋으로 관람하는 공연 등 특별한 오디오 퍼포먼스를 하고, 소리가 영상, 설치와 융합되는 새로운 예술형태의 가치를 추구한다. 공연이 시작되고, 헤드폰 속에서 남자와 여자의 목소리가 일상에 대해 이야기한다. "무슨 말을 하고 싶었던 걸까. 나는 알고 있을까. 무겁게 일상이라는 곳..."라는 조용한 나래이션이 옛 라디오드라마를 잠시 떠올리게 하였고, 매우 깨끗하고 선명한 좌, 우 패닝panning의 원근감을 조절한 남녀 목소리의 일상 속으로 관객들은 이내 빠져들었다. 장면이 바뀔때마다 길거리의 자동차소리, 커피숍의 사람들 소리 등 특유의 사운드 스케이프가 장면의 현장감을 살렸다. 여자가 말한다, "미안해". 남자가 말한다, "뭐가". 이 둘의 말이 독백인 듯 대화인 듯 서로 이어지기도 하고 아닌 것 같기도 하다. 창문 하나가 붉게 빛나다가 노랗게 빛난다. "밖에서 본 거울은 사각형이었다." 하얀 빛의 사각테이블 위로 붉은 빛의 십자가모양 조명이 비추인다. 여자가 "세상은 날 밀어내. 그러다 마침내 날 무너뜨리고 말 거야", "너에게 했던 말들, 그래, 그건 다 나에게 했던 말들이었어". 남녀 각자의 삶의 방식과 생각, 실로폰과 벨 소리 등이 세 개의 서로 다른 리듬으로 겹쳐져 반복하고, 사각형의 창에는 도시 속 사람들의 그림자가 다양한 포즈로 비춰진다. 희고 동그란 조명 하나가 천천히 움직인다. 헤드폰을 낀 남자배우가 의자에 걸터앉는다. "내가 찾던 사람은 나를 떠나갔다." 헤드폰의 대화 내용은 반복적이지만 점층적으로 겹치고 간격이 좁혀지면서 새로운 장면으로 접어든다. 여자가 "걷는다"라고 말하고 잠시 후 남자가 "걷는다"라고 말하니 배우가 걷는다. 여기서 조명(조명디자인: 노명준)이 중요한데, 천장에서 바닥까지 긴 일직선, 십자가, 네모 등 다양한 모양을 비추고, 공간의 바깥쪽에서 안쪽으로는 파란색, 흰색 노란색 붉은색 등 갖가지 색이 텍스트의 흐름에 맞추어 긴박한 장면전환 효과를 낸다. "흘러간다." 붉은 석양 노을같은 일직선 빛이 테이블을 통과하며 "세상이 날 밀어내"라고 여자가 말한다. 배우가 빛 통로를 어루만진다. 의식을 환기하는 종소리가 갖가지 높이와 음색으로 지속되고, 테이블에 번쩍이는 조명이 몇 번 비춰지더니, 여러 개의 네모 조명 회전하며 초록, 회색빛으로 뒤섞여 여러 곳을 비춘다. 창문 하나가 녹색으로 빛난다. "노래와 가로등 불빛만이 덮인 도시의 길." "난 이제 행복을 느끼지 않아. 불행도 느끼지 않아. 걷는다. 그걸로 만족해." 초록 조명 밑 밝은 음악과 카페 안 사람들의 말소리와 섞인다. 여자의 나레이션이다. "나의 기억 때문에 나는

이렇게 된 것일까? 누구를 사랑하지도 좋아하지도 말아야지. 클라이막스는 단조롭다." 어두움 속 테이블 옆에 흰 조명이 비춰지며 극은 끝난다.

작품의 제작단계에서 '아주 작은 창문'이라는 소제목으로 시작했던 남승연의 《장면(場面)》은 일상에서의 소통과 단절을 소재로 삼았다. 무대(무대디자인 정승준).는 작은 창문 여러 개로 이루어진, 사면이 막혀 둘러싸인 정육면체 구조물이다. 기승전결의 구조가 아니라, 대사가 이어지고 끊어지는 것을 반복하는 방식으로 주제인 '소통의 부재'를 표현했다. 그래서, 텍스트(작: 윤성호)가 이 작품에서 중요한 위치를 차지한다. 창에 대고 말을 하는 것처럼 가까이 혹은 멀리 들리는 남녀의 목소리는, 사실 둘의 대화라기보다 각자의 독백일 수 있다. 생각은 연결되다가도 단절되고, 그렇게 기억으로 옮겨진다. 이러한 과정을 남승연은 사운드의 원근감을 통한 깊이와 거리, 좌우 패닝의 방향, 이퀄라이징EQ를 통한 공간의 현장 위에서 대사, 인물 사이의 거리, 생각의 층위를 표현하고 공감을 이끌어냈다. 총제적으로 극의 부분으로서 관객도 공간 안에 배우와 똑같은 모습으로 헤드폰을 끼고 오브제가 되고, 배우가 움직이는 커피숍과 길거리의 일부가 되면서 극에 동참하거나 지켜본다. 제목이 '장면'이었던 바, 장면 전환 시 조명과 소리의 변화와 맞물림이 한치의 오차도 없는 것이 관건이었는데, 실제 그 역할로 인한 생동감이 여실히 증명되었다. 공연이 헤드폰 속 배우 두 명의 나래이션과 무대 위 배우 한 명에 의해서만 진행되고, 그 신호cue가 될 수 있는 것은 조명과 헤드폰 속 음악, 사운드 스케이프 뿐이었기 때문이다. 인터랙션, 모듈레이션 등의 화려한 사운드 테크놀로지의 수식 없이, 오로지 목소리와 대화의 내용, 소리의 깊이감과 방향감만 가지고 이토록 충실한 사운드극을 만드는 것에 실로 놀라웠다.

일상적 소재에 대해 남승연은 "나와 동떨어져 멀리 있는 소위 말하는 '예술'이 아닌 정말로 내가 느끼고 공감하는 소재를 찾고, 일반적인 형태와 다른 형태의 공연을 추구하고자 했다"면서 "소통의 부재를 주제로 '창'의 안과 밖으로 '일상의 틀'을 이등분하고, 쳇바퀴처럼 반복되는 일상을 소리와 무대요소들로 표현했다. 관객은 창 안에 들어가게 된다. 공간을 마련해놓고 관객을 초대했을 때 비로소 우리의 공연이 완성된다"라고 말했다. 나의 일상, 나의 생각과 같아서 누구나 공감 가는 공연, 내가 사는 지루하고 반복되는 '일상'과 한 차원 높은 '예술'로의 탈피라는 이분법이 아니라, 그 둘이 함께 손잡고 갈 수 있다는 발상의 전환을 깊이있는 철학으로 보여준 무대였다.

김태원 《새로운 유형의 공연/속삭임의 회랑》, 안 보이는 보물소리 찾기 대탐사

김태원의 《새로운 유형의 공연/속삭임의 회랑: 작은 소리도 멀—리-까지 들리게 만든 회랑》은 은평구립도서관에서 1월 25일부터 31일까지 공연되었다. 보통의 음악작품은 외부에서 만들어 온 작품을 공연홀이나 전시관에서 스피커를 통해 듣게 되는데, 김태원은 책을 읽고 공부하는 도서관, '소리'를 내지 않거나 작게 내야 하는 곳에서, 사람들의 행위로 발생하는 미세한 소리를 채집하고 연구해 작은 스피커의 공연으로 옮겼다는 것에 큰 의미가 있다. 도서관 지하1층, 지상1, 2, 3층 안내데스크, 종합자료실, 서고 등에 전체 다섯 개의 장Scene, 열여덟 개의 작품을 공연했다. 다섯 장은 모두 '번역의 방'에 차례대로 번호가 붙은 제목이었는데, 김태원은 유망예술지원사업2013부터 아야프2014까지 '번역기'에 관한 작업을 계속해오고 있는 바, 도서관 역시 무음의 소리를 통해 끊임없이 사용자의 수많은 욕망과 욕구를 번역하는 거대한 번역기로 보았다. x가 y로 변환, 해석되는 수식구조처럼, 도서관에서도 수많은 사람들의 욕구와 바람이 각종 '서적'을 통해 해석되는 '번역' 과정을 거친다고 볼 수 있을 것이다. 필자는 '전시' 형태일 것인데 왜 '공연'이라 했는지 궁금해하며 1층 안내데스크의 <번역의 방1Scene1-기적의 책꽂이>부터 보았다. 작은 모니터에는 눈이 이곳, 저곳을 보며 깜빡이는 모습, 입술을 이따금씩 오므렸다 폈다 하는 모습이 클로즈업되어 보인다. 무엇을 의미하는 것일까 생각해보니 도서관에서 움직이지 않고 가만히 의자에 앉아 열심히 글을 읽는 눈동자와 입으로 들이마시고 내쉬는 숨이다. 창틀 위에 놓인 멀티채널 스피커에서는 책의 겉표지를 넘기거나 긁는 소리, 내지를 구기고 문지르는 소리 등 오로지 책으로 발생할 수 있는 열여섯 가지 녹음한 음원이 채널별 스피커로 분리되어 다양한 시간 간격으로 겹치면서 여러 가지 리듬과 다양한 선율을 만든다. 1층부터 "신기한 소리작업이다"라고 생각하며 어느 방을 구경할지 고민하던 중 운 좋게도 김태원 작가를 만나 직접 안내를 받으며 공연을 감상할 수 있었다. <번역의 방2-출판유통 전문가 초청 세미나>는 1층 종합자료실에서 공연중이었다. 빽빽한 서고 속에 지향성 스피커(지향성스피커 제작: 김지연)가 서고 높이 설치되어 있다. "키보드를 치는 소리Sound of typing on the keyboard, '탭 탭 탭tap tap tap.. ,' 종이 자르는 소리Sound of cutting paper,

'칙chik..'... ," 스태플러 찍는 소리, 문 여닫는 소리, 복사기 소리 등 종합자료실에서 발생하는 스무가지 소리의 종류를 말한 후 그 소리를 자신이 생각하는 의성어로 표현한다. 이것은 작가가 2013년 독일 프랑크푸르트 북페스티벌Book Festival에서 같은 방에 투숙한 외국인들을 상대로 의성어 리서치를 한 것이다. 이 소리는 항상 들리는 것이 아니라 청자가 스피커와 같은 일직선상에 있으면 들리고 스피커와 다른 방향이면 들리지 않는다. <번역의 방3Scene3>은 지하1층의 문화교실의 자바스크립트javascript로 웹 프로그래밍(이강일)된 소리번역기1과 소리번역기2의 작품이다. 소리번역기1은 2014년 은평구립도서관 1, 2층 계단, 문 통과하는 소리, 식당 접시, 손 소독기, 엘리베이터, 반납기, 좌석 예약, 스캔하는 소리 등 도서관 관련 서른 한 곳에서 사운드 스케이프를 녹음한 것으로, 관객이 아이패드 앞 마이크에서 하고 싶은 말을 하거나 소리를 내면, 이 서른 한 개의 소리 중의 하나로 컴퓨터가 번역해준다. 소리번역기2는 <번역의 방2>의 스무 가지 외국인의 의성어 음원을 라이브러리화 한 것인데, 관객이 직접 마우스로 선택하면서 사람마다 같은 소리에 대해 저마다의 의성어로 표현하는 것을 들을 수 있다. <번역의방4Scene4-소리번역기3>는 지하1층 시청각장애인실 물품의 사운드 스케이프가 조도센서, 거리센서, 압력센서 등과 라즈베리파이Raspberry Pi(저소득층 어린이를 위한 초저가 컴퓨터), 퓨어데이터Pure Data(멀티미디어 프로그래밍 언어)로 프로그래밍(센서, 프로그래밍: 이강일)된 것을 들려주는 공연이었다. 입장하면서 문을 열면 연필 만지는 소리와 비디오 케이스 만지는 소리, 책장 앞에 서면 책장 넘기는 소리와 책 꺼내는 소리, 의자에 앉으면 의자 미는 소리가 센서 근처의 스피커에서 들린다. 공연이 진행되는 모든 장소Scene에는 군데군데 눈 깜빡거리기, 입술 오므리기(실제로는 숨쉬기, 이중모음, 겹받침 단어를 발음하는 영상), 고개를 한쪽 방향으로 돌리기의 영상들이 한 두개씩 전시되어 있었다.

김태원의 작업에서 중요한 또 한가지는 팜플렛이었다. 이는 도서관 내부 공연이라는 '항해'를 안내하는 지도이자 가상세계로 통하는 문이었다. 팜플렛에는 각 장의 사진 작품이 실려있는데, 증강현실 기술이 적용되어 디노마드 어플리케이션으로 사진을 스캔하면 각 장의 해당 자리에서 리서치한 사운드 스케이프가 핸드폰을 통해 흘러나온다. 또한, 장마다 각 층의 평면도 위에 '가상 스피커 지도'를 표시한 것이 특이했다. 왜 '가상'인지 작가에게 물었더니 "실제 작업이 아니라 가상이며 허구다"라고 설명했다. 도서관이 소리에 대한 의지를 가진 하나의 커다란 스피커라면, 실제 소리 공연을 펼친 곳 이외의 장소에서도 소리를 낼 것이고 이를 듣기 위해서는 '가상스피커'가 필요하다는 것이다. 발음에 관한 작업도 있었다. <4장Scene4>에서는 말과 말: 밤과 밤:과 같이 음 길이에 따라 뜻이 다른 '동음이의어', <5장Scene5>에서는 ㅑ, ㅘ, ㅢ, ㅖ와 같이 처음과 끝의 입 모양이 달라지는 '이중모음', 닭-, 밝-, 맑- 처럼 어떤 조사나 접미사가 따르느냐에 따라 발음이 달라지는 '겹받침소리'의 목록을 실었다. 음악에서 음 길이와 음색, 프레이징 등의 차이가 모여 다른 결과물을 만들듯, 이 작업들은 도서관의 책을 구성하는 '언어'의 속성도 소리의 길이와 발음, 문맥 등이 조합하여 텍스트의 의미를 바꾼다는 것을 일깨워주었다. 또 한 가지 주시한 것은 큐시트였다. 작가는 눈 깜빡이기(1장), 코로 숨들이마시고 내쉬기(2,3장), 소리가 들리는 쪽으로 머리 향하기(4장), 소리를 내지 않고 동음이의어 말하기(5장)처럼 '소리 나지 않는' 퍼포먼스를 도서관 지하1층부터 지상, 옥상까지 시청각실, 정기간행물실 등 총 열 다섯 곳에서 시간 순으로 진행했는데, 그에 대한 각 장의 큐시트 다섯 개가 팜플렛에 실려있다. 작가는 도서관 이용객이 책상에 앉아서 하는 제한된 행위들을 면밀히 관찰하고, 이를 도서관이라는 공연장에서 관객이라는 배우들이 큐시트에 의해 행하는 퍼포먼스로 간주했다. 작가가 스스로 퍼포먼스를 큐시트에 의해 실행하고 그 결과물을 각 장 별 장소에 놓인 영상작업으로 기록한 것이다. 필자는 각 장의 가상스피커 지도 아래에 "듣는자세(쓰기운동)"라고 쓰여 있는 것이 의아했었는데, 큐시트의 퍼포먼스 신호로서 이해할 수 있다. 즉, 도서관 이용객의 가장 전형적인 행위로서 책을 '읽는다', 글을 '쓴다'는 것은 '해독'과 '번역'의 작업을 통해 이를 의미화하는 것이며, 소리와 관련해서는 보이지 않는 '소리'를 머리와 생각을 통해 '듣고' 있는 것과 같은 의미인 것이다. 본 공연의 부제 '작은 소리도 멀—리-까지 들리게 만든 회랑'처럼, 실제 크게 들리거나 작게 들리거나 마음속으로 들리거나, 모두 '읽고 쓰는 것'은, '들리고' 있는 것이다. 작가는 이들이 비록 의자에 앉아서 읽고 쓰고 듣는 정적인 자세일지라도 그 행위를 통해 작가의 거대한 공연에 참여하고 있다고 해석했고, 그 정적인 퍼포먼스 몇 가지를 팜플렛에 작성한 것이다.

김태원은 "소리를 형식적, 내용적으로 어떻게 확장시키느냐의 문제에 대해, 소리가 금기시되는 공간인 도서관에서 역설적으로 그 확장의 실마리와 동력을 얻게 된다"라며, "'장소 특정적site-specific' 공연에 관심이 있었다. 가장 소리가 없

는 공간에서 소리를 찾아내는 과정은 결국 이미 모든 곳에는 소리가 유형이나 무형으로 존재할 것이라는 가정에서 시작해 그것을 증명하는 일련의 과정이었다"고 말했다. 이처럼 《속삭임의 회랑》에서는 소리기술의 현란함보다 패러다임의 전환과 사고의 확장을 통해 소리를 재발견하고 '팜플렛'이라는 보물찾기 지도를 가지고 우리가 지나쳤던 스피커를, 미지의 소리를, 꿈틀거리는 소리의 의지를 찾아나서는 대탐사와 항해를 할 수 있었다. 작가의 새로운 생각과 개념에 공감하며 적극적으로 참여하는 열린 의지를 가진 자만이 그런 특혜를 누릴 수 있음은 당연하다.

사운드 신세계, 테크놀로지와 사운드 '신개념'의 결합

아야프 2014 사운드, 전자음악 분야의 작곡가와 작가 다섯 명의 작품에서 소리와 음악을 다루는 방식은 서로 다르지만, '공연'이라는 목표를 향해 자신의 의미와 개성으로 일구어낸 과정을 거쳤다는 공통점이 있었다. 김태원은 공연장이 아닌 도서관 속 미세한 소리, 잘 들리는 않는 소리까지 범위를 확대해 연구하고 전시와 공연의 탈 경계를 이루어내었고, 센서와 증강현실, 멀티채널, 지향성스피커 제작기술 등을 활용했다. 남승연은 일상을 재발견해 온전히 말과 소리로만 공연을 구성하되 스피커가 아닌 헤드셋으로 선명한 원근감을 이용하여 특화된 공간감을 구현했다. 일상의 사운드스케이프로 새로운 공연형태를 만들어낸 것에서 남승연과 김태원은 연관된다. 남상봉은 전자악기인 엠포이를 직접 개발하고 이것을 필두로 자신만의 현대음악 작품과 라이브 일렉트로 어쿠스틱 작품을 '밤'이라는 주제로 통합해 대담한 음악극을 펼쳤다. 조진옥은 전통 판소리 수궁가에 24채널 입체음향을 통한 소리의 공간화spatialization와 수묵담채영상이라는 옷을 입혔다. 이동규는 음악과 음향을 기반으로 지역 전통의 이야기에 키넥트, 모래그림, 무용, 영상의 멀티미디어로 청중에게 마법과 같은 주문을 걸었다. 조진옥과 이동규는 공통적으로 한국전통소재에 멀티미디어 기술을 세련되게 활용했다. 각 작가들은 '공연'을 규정하는 의의, 공연에서 보여주고자 하는 목표, 음향과 음악을 다루는 방법이 달랐지만, 다섯 명의 예술가 모두 자신이 이루고자 하는 개념을 위해 새로운 방법론으로 소리에 접근했다. 앞으로 음향 및 전자음악 작업이 예술가와 관객에게 더욱 매력적인 기회를 제공하기를, 젊은 예술가들은 새로운 공연형태를 개척하고, 직관적인 감상이 가능하고 소리의 가치가 충분히 전달되는 작품들이 많이 탄생되기를 바란다. 아야프2014에 참여한 신진예술가은 이구동성으로, 신진으로서의 패기 넘치는 이 열정의 시기에 아야프 사업이 아니었다면, 시행착오를 거쳐 깊이있는 고민과 탄탄한 과정위에서 그토록 자신이 구상한 세계를 실행하지 못했을 것이라고 말했다. 젊은 창작자들에게 마음껏 새로운 실험을 가능하게 해준 아야프사업이 많은 이들의 관심과 함께 더욱 활성화되어, 기발하고 참신한 모험과 도전으로 예술세계와 음악세계가 더욱 확장되기를 바라는 마음이다.

A Review of *Seoul International Computer Music Festival 2015*

Son, Min-jung

Seoul International Computer Music Festival 2015 was held at the *Jayu* Theater of Seoul Arts Center for four days from September 30th to October 3rd. About 170 pieces from all over the world were applied for the concerts, and all told, the thirty selected works by musicians from twelve countries were put on stage. The Ensemble *Unitedberlin* was invited to the last concert of the festival for the grand finale. The program of the festival encompassed various forms of computer music such as 4-8 channel tapes, live electronics, electro-acoustics, and audio-visual performances. All the times of the four-day *SICMF 2015* was under the attention of a diverse audience and many related musicians, came to a successful close, as this event is generally regarded as one of the most notable and prestigious events in the field of computer music.

서울국제컴퓨터음악제 2015 참관기

손민정

2015년 9월 30일 수요일부터 10월 3일 토요일까지 나흘 간 예술의전당 자유소극장에서 서울국제컴퓨터음악제(이하 컴퓨터음악제)2015 Seoul International Computer Music Festival(SICMF)가 개최되었다. 백칠십 여 응모작 중에서 당선된 서른 작품이 연주되었으며 앙상블 유나이티드베를린Ensemble Unitedberlin이 피날레 공연을 담당했다. 열 두 개국의 작곡가들이 참가하였으며 평소에는 접하기 힘든 폭넓고 다양한 형식의 컴퓨터음악을 한 자리에서 체험할 수 있었다. 공교롭게도, 국제컴퓨터음악학술제International Computer Music Conference(ICMC)가 미국 텍사스에서 동시에 개최되고 있는 상황이어서 적지 않은 우려를 하였음에도 빈자리가 보이지 않을 만큼 성황리에 진행되었다. '컴퓨터음악'이라는 매우 구체적이고도 특정한 장르에 관한 세계적으로도 흔치 않은 규모의 음악제이기에 관중들 역시 대단한 기대를 가진 듯 보였다. 전체적인 페스티벌에 관한 리뷰를 풀어놓은 후, 나흘 간의 콘서트에 올려진 작품에 대하여 차례로 다루겠다.

1. 페스티벌 리뷰

작품선정 현황

1994년부터 시작된 컴퓨터음악제는 공모를 통해 작품이 취합되며, 선정 기준은 예술적 가치가 우선이되 실행 가능 여부도 고려된다. 컴퓨터음악제는 2000년대 초반을 기점으로 국제적인 주목을 받기 시작했으며, 올해에는 170여 작품이 응모하였다. 초창기에는 전자음악연구소가 발달된 서유럽의 작곡가가 주로 참가하였지만, 최근에는 미주지역 및 동아시아 작곡가들이 적극적으로 참가하고 있는 추세이다. 컴퓨터음악의 세계적 확산을 의미한다고 생각할 수 있겠다.

컴퓨터음악제2015에 선정된 작곡가 서른 명의 출신 지역을 살펴보자면, 한국이 열두 명, 유럽이 일곱 명, 캐나다 및 미국이 다섯 명, 중국을 비롯한 동아시아가 다섯 명, 뉴질랜드가 한 명이다. 총 열두 개 국의 작곡가들이 참가하였으며 이는 컴퓨터음악제의 국제페스티벌로서의 면모를 역력히 보여주었다고 평가할 수 있다. 한국을 제외한다면 캐나다, 독일, 일본이 각각 세 명으로 가장 많이 선정된 국가이다. 실행위원 이은화와 김범기에 따르면, 대부분의 참가 작곡가들은 컴퓨터음악연구소 또는 전자음악연구소에 소속되어 있으며 최근에는 중국 작곡가들의 관심이 커지고 있다고 한다.

현장

실험적인 무대를 연출하기에 적합한 예술의전당 자유소극장에서 공연되었다. 컴퓨터음악 공연에 있어 중요한 요소 중 하나는 스피커의 공간적 배치인데 무대의 좌우에 2개의 메인 스피커를, 중앙 좌우 및 후방에 여섯 개의 스피커를 배치하여 총 8개의 스피커를 통한 '소리의 공간화'를 꾀한 음향 시스템을 경험할 수 있었다. 아울러, 비교적 낮은 무대 덕분에 관객의 작품에 대한 몰입도 수월했다. 비디오를 동반하는 작품을 위하여 무대 배경에는 스크린을 내려놓았으며 무대와 객석 사이에는 컴퓨터 및 기기를 배치시켜 진행위원과 작곡가들이 무대를 바라보며 연주를 진행하였다.

참여 작곡가 전현석은 타 음악페스티벌과 구별되는 컴퓨터음악제의 특성으로 감상과 체험의 차이라 말한다. 컴퓨터음악제에서는 청각과 더불어 시각과 촉각을, 시간적 흐름을 비롯한 공간적 확대를 이용한 실험적인 작품을 무대에 올리기에 관객은 다각적인 '체험'을 할 수 있다는 말이다. 강력한 스피커 음량과 강렬한 스크린의 불빛은 단순한 청각적, 시각적 한계를 넘어서기 때문이다. 아쉬운 점으로는 '테이프 연주'만 있는 경우에는 대부분의 조명을 끈 상태에서 진행하게 되는데 비상구, 컴퓨터의 스크린 불빛 등의 의도되지 않은 몇 개의 불빛이 몰입에 방해요소로 작용하였다는 것이다.

프로그램 구성

두 시간 남짓의 프로그램이 나흘 연속 연주되었다. 우선, 공연형식에 따라 작품의 성격은 크게 두 가지로 나눌 수 있는데 첫째는 고정매체fixed-media 또는 테이프tape 라 불리는 형태로 연주될 음악이 미리 저장되어 재생되는 경우이며, 둘째는 라이브 일렉트로닉스live electronics형식으로 작곡가가 직접 연주에 참가하여 실시간으로 소리를 조작해가며 음악을 만들어가는 경우이다. 컴퓨터음악제2015에서는 이 둘이 거의 동등한 비율로 구성되었다. 악기연주가 동반되지 않은 채 테이프만 연주되는 작품이 여섯이 있었고, 라이브 일렉트로닉스의 경우에는 모두 악기와 함께 연주되었다. 앞의 두 가지 편성과 다른 작품으로는, 비디오가 동반되는 여섯 작품과 퍼포먼스를 포함한 두 작품이 있었다.

프로그램의 순서는 대체로 무대전환의 효율성을 고려하여 정해진 것으로 보인다. 따라서, 다른 페스티벌이 일반적으로 갖는 소주제와 같은 '주제적 통일성'이 부족하다는 면이 아쉽다. 다만, 매 콘서트의 프로그램은 가능한 다채로운 형식을 포함하도록 구성되었는데. 예를 들어 비디오 및 퍼포먼스가 동반된 일곱 편의 작품은 나흘에 걸쳐 골고루 분포되어 각 콘서트가 개별적 완결성 및 독립성을 갖도록 하였다.

작곡가의 철학

컴퓨터 음악은 작곡가의 철학에 따라 작품의 성격이 극명하게 갈린다. 음악작품의 논리와 구성에 역점을 두는 작곡가인 경우에는 컴퓨터를 하나의 악기로 취급할 뿐, 지나친 효과를 자제하는 경향을 띤다. 그런가 하면, 총체적인 체험을 도출하려 하거나 특정 사회적 이슈 또는 개인적 상황을 주제화 시키려는 경우에는 컴퓨터음악의 음향적 요소를 부각시키기 마련이다. 컴퓨터음악제2015에서는 유태선, 이병무, 임승혁의 경우가 전자에 속하며, 남상봉을 비롯한 비디오적 요소를 포함하는 대부분의 작곡가들이 후자에 속한다. 청자의 입장에서 구별하자면, 전자의 경우에는 대부분이 어쿠스틱 악기연주를 부각시키거나 음조직의 구성에 집중한 테이프연주를 만들어내며, 후자의 경우에는 막강한 음량과 자극적인 시각적 효과를 과감히 사용한다고 분석할 수 있다.

컴퓨터음악 작곡과정의 특수성에 관한 작곡가의 경험은 이렇다. 참여 작곡가 이원에 따르면, 컴퓨터음악은 타 장르와는 달리 작곡에 앞서 '소리' 자체를 만들어 내기 때문에 그 창조 과정 속에서 또 다른 희열을 느낀다고 한다. 물리학 전공자가 작곡의 매력에 빠질 수 있었던 이유는 컴퓨터음악이었기에 가능했으리라 짐작한다. 또 다른 참가 작곡가 유태선은 '일반적인 작곡에 비해 소리 재료의 만듦부터 시작하기 때문에 고려해야 되는 부분이 많으며, 소리가 어떻게 전달되는가에 관해서도 고민을 많이 한다'고 덧붙인다. 이는 소리의 창조과정을 비롯하여 음악의 음향적 요소가 중요하다는 것을 방증하며, 얼마나 창의적인 소리를 만들어내는지, 어떻게 총체적인 체험을 이끌어내는지가 중요하다는 결론이다.

2. 작품 리뷰

음악회 1

첫 곡은 중국출신 작곡가 치우샤오 리Qiuxiao Li의 클라리넷 독주와 테이프를 위한《우송, 호랑이와 싸우다Wu Son Fights the Tiger》이다. 소재는 수호전에서 따왔다. 우송은 워낙 힘이 장사라서 술에 취한 상태임에도 호랑이를 맨손으로 물리쳤다는 이야기다. 이 작품은 Jun Qian 박사의《동서의 만남East meets West》이라는 녹음 프로젝트를 위해 만들어졌다고 한다. 중국의 설화를 서양 현대음악 어법에 맞추어 새롭게 만들어내고자 했던 것이다. 클라리넷의 넓은 음역과 다채로운 음색을 충분히 활용하여 우송의 호기로운 성격을 표현해 내는가 하면, 전자음향의 타악기적 소리는 호랑이와의 싸움에서 느낄 수 있는 긴장감을 표현했다. 작곡가의 의도대로 베이징 오페라의 요소가 효율적으로 사용되었으며 컴퓨터음악제2015의 첫 작품으로 적절하게 프로그램되었다.

두 번째 곡은 캐나다 작곡가 데이빗 브레잔David Berezan의 5.1채널 테이프를 위한 《등대선Lightvessels》이다. 바다의 음향세계Maritime Soundworlds를 탐색하려는 연작 중《부표Buoy》와《계류기구Moorings》에 이은 세 번째 작품이다. 테이프로 만들어진 작품이기에 조명을 끄고서 감상하게 되는데 마치 바다에 온 것 같은 느낌을 생생하게 연출해 내었다. 단, 바닷소리, 등대선의 기계소리, 등대선의 움직임, 등대선 주변의 각종 음향을 사실적으로 잘 표현하였다는 것 이외에는 작곡가의 독창적인 음악적 구성력 또는 논리를 찾기는 힘들었다는 점이 아쉽다.

세 번째 곡은 미국 작곡가 앤드류 시거 콜Andrew Seager Cole의 네오 색소폰 사중주와 4채널 라이브 일렉트로닉스를 위한《아로-드림스케이프Aro Dreamscape》이다. 이 작품에서는 '소리의 공간화'를 유감없이 체험할 수 있었다. 작곡가는 뉴질랜드 아로 밸리Aro Valley에 머무를 때 침실 창에서 몽롱한 상태로 새소리 녹음을 하게 되는데, 이 작품의 착상이 되었다고 한다. 그러나 투이Tui라는 새의 소리만 바리톤 솔로를 통해 구체적으로 표현될 뿐, 나머지 새소리는 추상적인 형태로 바뀐다. 우연성aleatoric system에 의거해서 새소리의 횟수와 오버랩을 탈규칙적으로 만들어내는가 하면, 관객이 산 속을 걸어가는 체험을 느낄 수 있도록 각 색소폰들은 무대에, 관객석 왼쪽 2층에, 관객석 오른쪽 2층에, 관객석 후방 2층에 개별적으로 배치되어 소리를 입체화시켰다. 청각의 시각화는 '꿈풍경dreamscape'이라는 제목에서 그 의도를 읽을 수 있으며 흥미롭게 체험할 수 있는 작품이었다.

전반부 마지막 작품으로 한국 작곡가 박한나의 플루트, 베이스 플루트, 라이브 일렉트로닉스를 위한《5》가 연주되었다. 숫자 '5'가 주제로 사용되었는데 소리를 최소 단위로 해체시켜 다섯 가지 패턴으로 이루어진 새로운 음형들을 만들어내었다. 음악 외적인 소재를 다루기보다 음악자체의 논리에 역점을 둔 작품으로서 플루트의 긴장된 호흡이 특징적이었다.

후반부 시작 작품은 미국 작곡가 존 니콜스 3세John Nichols III의 8채널 테이프를 위한《게이트Gates》였다. 전반부 마지막 작품에서 자아낸 조심스러운 음악진행의 긴장감을 단순에 날려버리기라도 하듯 강렬하고 다이내믹한 사운드를 만들어냈다. 컴퓨터음악제2015의 기획자가 의도했던 바였는지는 알 수 없으나 페스티벌 첫 날 프로그램에서 가장 잘 된 구성이었다고 여겨진다. 작품의 착안점은 플레이아데스 성단 별자리Pleiades constellation이며 작품의 중반과 마지막에서 음악적으로 묘사되었다고 한다. 작곡가의 해설에 따르면 이 작품은 다분히 종교적인 관점을 포함하고 있는데 사운드의 지속적인 펼침에서 신비로움을 느낄 수 있었다.

여섯 번째 곡은 뉴질랜드 작곡가 클로비스 맥이보이Clovis McEvoy의 테너 리코더와 8채널 라이브 일렉트로닉스를 위한《합류Conflux》이다. 제목에서 알 수 있듯이 테너 리코더는 뉴질랜드 원주민의 자연 유산을 의미하며 전자음향은 산업화와 도시화를 말한다. 연주방식에 있어서 테너 리코더 연주자가 철저한 주도자가 되었으며 전자음향은 자연스럽게 연주자의 '템포', '다이내믹스', '음선택'에 적응하며 때로는 충돌하고 때로는 화합하면서 역동적인 관계를 만들어가려 했다. 자연과 과학의 조화를 꿈꾸려는 작곡가의 의도에 비해, 과학 즉 전자음향의 역할이 다소 미약하다는 느낌이 든다.

일곱 번째 작품은 한국 작곡가 남상봉의 엠포이mPoi 독주를 위한《쥐불》이었다. 엠포이는 남상봉이 직접 개발한 전자악기로 원의 궤적이 센서를 통해 컴퓨터로 전달되면 이 동작에 따라 다양한 소리와 빛을 만들어낸다. 컴퓨터음악제2015 첫 날의 하이라이트라 여겨질 만큼 청중의 반응이 좋았다. 무대에는 흰색 체조복을 입은 한 남성이 쥐불을 연상케 하는 기구를 들고 나와서 화려하고 다채로운 불빛 원형 궤적들을 만들어낸다. 이 궤적들은 움직임에 따라 빨강, 파랑, 초록, 노랑 등의 불빛으로

변환되고 음악 역시 변환된다. 컴퓨터음악이 '소리는 빛'이 될 수 있다는 원리에서 시작되었다면, 한 발 나아가 '움직임이 소리와 빛'이 될 수 있도록 만든 것이다.

마지막 작품은 일본 작곡가 히데아키 이소베Hideaki Isobe의 기타와 라이브 일렉트로닉스를 위한 《오크Oak》였다. 시종일관 '호흡'에 집중되었던 음향이 마지막 곡에서 '현의 떨림'으로 전환되었다. 기타의 소리는 오크나무의 몸통을, 전자음향은 그 몸에서 나온 나뭇가지를 표현하였다. 특정한 몇 개의 소리를 중요하게 다루며 반복하는 모습을 보이며, 배음을 중요하게 다루는 것은 몸통과 나뭇가지의 연결고리를 연상케 한다.

음악회 2

처음과 마지막 공연이 비주얼을 동반한 작품이었으며, 첫 번째 작품의 비디오가 아날로그적 영상이었다면 마지막 작품의 영상은 빛이 되어버렸다. 공연된 순서대로 풀어보겠다. 첫 번째 작품은 박순영의 바이올린, 비올라, 비디오, 행위예술, 그리고 일렉트로닉스를 위한 《북아현코스모스-1악장》이었다. 작가는 자신이 사는 북아현동이 아파트건설로 인하여 나날이 황폐해지고 있지만, 그럼에도 불구하고 여전히 변하지 않는 우주자연의 법칙이 있고 사계절의 순환이 존재한다는 것을 표현하고자 했다. 무대 한가운데에 넓은 천이 뭔가를 덮고 있다. 생명체의 에너지가 아닐까 싶지만 작품이 한참 동안 진행될 때까지도 꼼짝을 하지 않는다. 스크린에서는 쓸쓸하고 적막한 북아현동 폐허에서 바이올리니스트가 황량하게 뭔가를 연주하는 장면이 보인다. 무대에서는 똑같은 연주자가 또 다른 알 수 없는 선율을 연주한다. 작품은 점점 격정적인 형태로 발전하며 급기야 무대 가운데에서 꿈틀거리고 있던 헝겊아래에서 생명체로 보이는 행위예술가가 격정적인 몸짓으로 우주의 진동을 전달한다. 감각적이면서도 구체적인 오디오 비주얼 예술의 형식을 빌어 현실에서 부딪친 문제와 그로 인한 고통을 효과적으로 풀어낸 작품이다.

두 번째 작품은 독일 작곡가 루드비히 베르거Ludwig Berger의 테이프를 위한 《가상적 섬의 1:1 스케일 지도Mapping of hypothetical islands on the scale of 1:1》이었다. 철저히 추상적인 개념을 표현하려 했으며, 지시적이거나 묘사적인 작품은 아니기에 구체적이고 보편적인 이해를 요구하지도 않는다. 경우에 따라서는 목적과 수단이 전도되기도 하는데 이 작품의 경우에는 작품의 펼침 자체가 목적임과 동시에 수단이었다고 생각된다.

세 번째 작품은 이탈리아 작곡가 발리오 드 보니스Valeo De Bonis의 바이올린과 4채널 테이프를 위한 《내 몸 안의 고통과 같이Like a pang in my body》이었다. '팽pang'은 경련과 같은 자극적이면서도 급작스러운 고통을 의미한다. 시종일관 날카롭고 고통스러운 음향이 거친 바이올린의 소리와 함께 우리의 감각을 아프게 하는데, 이미 갖고 있는 통증이 있다면 오히려 그 통증을 잊도록 하려는 듯하다.

네 번째 작품은 캐나다 작곡가 제임스 오칼리간James O'Callaghan의 테이프를 위한 《몸-울림Bodies-Soundings》이었다. 제목에서 말하는 '몸'은 어쿠스틱 기타와 장난감 피아노의 인격화된 몸체를 말한다. 연주자는 없다. 청중이 듣는 것은 보이지 않는, 때로는 알 수도 없는 소리들의 진동과 얽힘을 느낄 뿐이다. 철저히 진동의 살아있음을 느끼도록 디자인되었다고 보인다.

다섯 번째 곡은 엄숙한 분위기의 작품으로 대만 작곡가 퀘이-판 린Kuei-Fan Lin의 소프라노와 테이프를 위한 《임마야Immayah》이었다. 원래 실내악과 일렉트로-어쿠스틱 음악을 위한 《삼위일체》의 3악장으로 '성령'을 묘사하려 한다. 전자음향의 상당한 부분이 소프라노의 녹음 샘플을 활용하면서 작품의 주제를 부각시키려 했다는 점에서 메시지가 명료하게 전달되었다고 느껴졌다. 성스럽고 종교적인 분위기를 인간의 목소리로 구체화시키려 했다는 것도 효율적인 발상이었다고 생각된다.

여섯 번째 작품은 이원의 테이프를 위한 《테이핑 임펄스Taping Impulses》였다. 물리학 전공자답게 소리의 발상과 전환에 호기심을 집중시켰다. 이원은 접착 비닐 테이프를 잡아떼는 소리에서 작품구상을 시작했다. 전혀 음악적 소리로 들리지도 않고, 그렇게 인식조차 하지 못하는 소리를 음악으로 전환시킬 수 있었는 것은 컴퓨터 프로그램의 놀라운 능력 덕이다. 게다가 마그네틱 테이프를 더 이상 사용하지 않게 된 테이프 음악을 위하여 접착용 테이프 소리를 사용하여 음악을 만들었다는 아이디어는 본질과 허상의 전복에서 느껴지는 허탈감을 상기시킨다.

일곱 번째 작품은 유태선의 생황과 8채널 라이브 일렉트로닉스를 위한 《가온》이었다. 이번 페스티벌에서 유일하게 국악기를 사용한 작품이었다. 연주방식은 매우 실험적인 방향으로 맞추는가 하면, 전체적인 작품의 방향은 어쿠스틱한 음악적 논리구성에 초점을 두면서 전자음향의 효과나 자극을 최소화시켰다. 덧붙여, 천지만물의 중심을 의미하는 '가온'을 작품의 제목으로 두면서 조화로운 소리를 찾으려는 동양음악철학을 구현하려 하였다. 서양과 동양, 고전과 현대의 절묘한 대치와 조화를 느낄 수 있었다.

마지막 작품으로 세르비아 보얀 밀로세빅Bojan Milosevic의 오디오 비주얼 연주를 위한 《에타르Etar》이었다. 이 날 무대에 올려진 작품 중 확실한 볼거리를 주었던 연주라 하겠다. 대형 스크린을 가득 메운 영상은 이미지가 아닌 '빛'에 가까웠다. 그 형체가 지나치게 자극적이고 강렬해서 눈을 뜨기가 힘들 지경이었다. 마치 감각의 한계를 시험하는 듯했다. 작가는 빛과 음향의 통합을 추구했으며 에너지와 영혼의 근원에 자리한 '핵'에 다가가고자 했다. 시종일관 긴장의 연속이었으며, 인간의 육체가 견디기에는 우주의 핵으로 다가가기가 쉽지 않음을 고통스러울 만큼 체험하게 되었다.

음악회 3

음악은 구체적이고도 직접적으로 체험할 수 있지만 여전히 추상적이기에 보이지 않는 많은 추상적인 개념들을 표현하려 노력한다. 이는 음악의 특권이기도 하다. 첫 번째 작품은 일본 타쿠토 푸쿠다Takuto Fukuda의 플루트, 바이올린, 하프, 8채널 일렉트로닉스를 위한 《원심의 선율Centrifugal melodies》이며, 원심력이라는 에너지를 음악으로 옮겼다. 역동적인 에너지에서 뿜어 나오는 긴장감을 극대화시키는 데에 일렉트로닉스를 효과적으로 활용하였다. 특히, 세 악기가 맡은 선율의 펼침은 다분히 우주적 이미지를 연상케 하기에 충분했다.

두 번째 작품 역시 일본 출신 작가의 작품으로 케이수케 야기사와Keisuke Yagisawa의 오디오 비주얼 미디어를 위한 《xtx to》이었다. 이번 작품은 '문체연습'과 '텍스트 읽기'의 변주를 주제로 한다. 프랑스 시인 레몽 크노Raymond Queneaus의 "문체연습Exercise de style"을 일본어로 번역한 글과 이것을 읽는 소리를 사용한 것이다. 흑백 스크린에서는 끊임없이 쏟아지는 일본어 글자의 해체와 조합을 이미지로 보게 되며, 그 이미지가 '음성'으로 읽혀진 소리들이 쉬지 않고 깨어지고 붙여지기를 반복하며 변주된다. 음악이라 말하기에는 부담스러울 정도로 음향적이며 음성적이다.

세 번째 작품은 오예민의 피아노, 라이브 비디오와 일렉트로닉스를 위한 《공감각의 순간Synesthetic Moment》이었다. '공감각synesthesia'은 천재 예술가들이 갖고 있다는 능력으로 한 가지의 감각적 자극이 다른 감각과 연결되는 현상을 말한다. 칸딘스키의 경우 음악을 들으면서 받은 청각적 자극이 동시에 시각적인 감각으로 전환되어 그림의 영감이 되었다고 한다. 작곡가 오예민은 음악과 영상의 조합을 통해서 공감각을 효율적으로 창출할 수 있는 환경을 만들고자 했다. 작품이 시작되면 피아니스트가 등장하며 그녀의 몸짓 하나하나가 스크린에 재현된다. 때로는 실제모습 그대로, 때로는 작가의 미학적 선택에 의해 해체되어 보여 진다. 청중은 그녀의 음악을 수동적으로 느낄 수 없으며 그녀의 몸짓을 보며 어떻게 음악화되는지를 시각적으로 체험하게 된다. 공감각을 활용한 기존의 작품들이 음악소리에서 연상되는 시각적 이미지에 초점을 맞춘 것에 반해 이번 작품은 음악행위 자체를 이미지화했다는 점에서 흥미로웠다.

전반부의 마지막 작품은 이병무의 4채널 테이프를 위한 《작은 창들Tiny Windows》이었다. 첫 번째 곡이 물리학적 개념과, 두 번째 작품이 음성과 글자, 세 번째 작품이 공감각적 이미지와 연결시켜 음악을 풀어보았다면 이번 작품은 철저히 음악구조 자체에 몰입하였다. '작은 창들'은 세상을 아주 작은 알갱이들grains로 분해하고 걸러내는 모종의 시스템들을 의미한다. 이병무가 선호하는 개념은 '해체'다. 기존의 관습적인 개념으로부터 벗어나 근원적이며 본질적인 진실에 다가가고자 한다. 사랑하는 아들의 일상적인 목소리를 알 수 없는 소리의 조각으로 해체시키기도 했다. 사실상 우리는 모두 어떠한 '창'으로든 세상을 자신만의 방식으로 바라보고 판단하고 있다. 이번 작품은 우리들의 일상을 왜곡시키고 해체시키고자 그 창을 다원화시키고 최소화시킨 것이다.

다섯 번째 작품은 폴란드 작곡가 니콜레 부진스카Nikolet Burzyńska의 플루트와 4채널 라이브 일렉트로닉스를 위한 《플렉트로Flectro》였다. 어쿠스틱과 일렉트로닉스를 혼합하여 창작하는 경우 어쿠스틱과 일렉트로닉스 중 어떤 악기에 무게를 두는가에 따라 작품의 성격이 크게 갈려진다. 제목에서 알 수 있듯이, 이 작품은 그 두 소리의 통합을 목표로 한다.

이번 페스티벌의 특징인지, 컴퓨터음악의 전반적인 경향인지는 면밀한 연구를 해 보아야 알겠지만, 컴퓨터음악에 목관악기를 활용하는 경우가 두드러진다. 추정컨대, 일렉트로닉스가 명시하는 기계성에 상반되는 인간성, 특히 인간의 호흡과 목관악기가 연관되기 때문이라 여겨진다. 이 작품에서는 기계성과 인간성의 유기적인 조합을 추구하려 한다.

여섯 번째 작품은 캐나다 작곡가 마리-헬렌 브롤트Marie-Hélèn Breault와 마틱 베다드Martic Bédard의 공동작으로 테이프를 위한 《복제Replica》이었다. 작곡가들이 밝혔듯이 이 작품은 순수 어쿠스마틱한 작품으로 기획되었기에 선율이나 리듬 등의 고전적인 음악적 요소를 찾기는 힘들며, 음색이나 스펙트럼을 위주로 만들어졌다. 어쿠스마틱acousmatic은 '들려지기 위한' 음악을 뜻한다. 철저히 라이브를 배제하였으며, 플루트의 소리를 녹음과 변환을 통하여 계획적으로 발전시켰다. 다소 평이하게 느껴질 수도 있으나 다채로운 전자음향의 기법을 노골적으로 드러낸다는 점에서 입문자에게는 유익할 수 있다고 생각된다.

일곱 번째 작품은 김한신의 오디오 비주얼 미디어를 위한 《다da》이었다. "다"는 '나는 있는 나다'를 의미하는 독일어 문장 'Ich bin der Ich-bin-da'의 일부에서 따왔다. 작가는 '있는'의 의미를 찾아가는 작업을 '사운드'와 '이미지 안에 있는 사운드'를 찾아가는 것으로 설정하여 작업했다고 한다. 스크린의 이미지는 화면이 끝없이 분할되어가다가 결국 원상태로 돌아오는 구성이다. 명료한 이미지와 비교적 구체적인 사운드의 결합이 작곡가의 의도를 이해하기에 충분했다.

콘서트 마지막 작품은 전현석의 메조 소프라노, 타악기와 테이프를 위한 《크로노스의 정원The Garden of Kronos》이었다. 컴퓨터음악의 '체험적 요소'를 감안하며 음향의 '전달'에 역점을 두었다. 8채널을 사용한 다이내믹한 전자음향이 있음에도 불구하고 어쿠스틱 타악기를 메인으로 등장시킨 것은 제목 그대로 '공간'적인 체험을 극대화하기 위한 것으로 여겨진다. 크로노스의 광기 어린 분노와 불안감을 개인의 내면적인 상상이 아닌 시각적이면서도 공간적인 분위기 연출로 표현하려 했다는 것이 참신했다.

음악회 4

마지막 날에는 여섯 곡으로 구성되었으며 전 곡을 앙상블 유나이티드베를린이 연주하였다. 상이한 작곡가의 작품들을 시종일관 동일한 연주자들이 연주한다는 것은 청중으로 하여금 '작품' 자체에 몰입할 수 있도록 도와주었다. 컴퓨터음악의 범주가 워낙 넓은데다가 연주자의 성향마저 다를 경우에는 그 변수가 상상할 수 없을 만큼 커지기 때문이다. 이 날 작품의 편성에 있어서 관악기가 한 번도 등장하지 않은 것도 특이했으며, 현악기 위주로 편성된 작품들에서 '논리'적인 성향을 강하게 느꼈다.

첫 번째 작품은 조영미의 바이올린, 첼로, 일렉트로닉스를 위한 《유전적 캐논Genetic Canon》이었다. 전통적인 음악어법인 캐논을 도입하되 생명과학의 한 분야인 유전학에서 차용한 자연선택, 교차, 변이 등의 개념들을 활용하였다. 독립된 선율을 반복하고 모방하되 다양한 변형들을 형성해 가며 변화무쌍하게 진행시키며, 그리고 내재된 유전자가 다음 세대에 영향을 미치듯 그 선율들을 '진화'시키려 했다. 선율이 마치 유기체가 되어 꿈틀거리는 것 같았다. 다소 복잡하게 느껴질 수는 있으나 다이내믹한 전자음향에서 선율의 생명력을 느낄 수 있었다.

두 번째 작품은 네덜란드 작가 미셸 반 데 아Michel van der Aa의 피아노와 비디오를 위한 《트렌시트Transit》이었다. 스크린에서는 한 노인이 등장한다. 알 수 없는 강박증으로 집 밖을 나가려 해도 도저히 문을 열 수가 없다. 어쩌면 자신의 내면에 있는 벽을 넘어서지 못하는 것일지도 모른다. 노인의 일상은 집안을 배회하는 것이며, 극심한 외로움을 견디기 힘들어한다. 피아노의 소리는 미리 녹음되어 재생되는 전자음향과 충돌하기도 하면서 이야기를 소리화 한다. 전자음향이 갖는 건조함이 노인의 어찌할 수 없는 무기력함을 효과적으로 표현했다.

전반부 마지막 작품으로 임승혁의 바이올린, 비올라, 타악기, 피아노, 테이프를 위한 《가기》가 연주되었다. 이번 페스티벌에서 가장 거대한 편성을 가진 작품이다. 그러나 주제는 가장 심플하다. 논리와 직관에 의한 음악의 '가기'이다. 몇 해 전부터 임승혁은 논리와 직관에 대하여 여러 차례 이야기를 한 적이 있다. 이번 작품을 접하는 순간, 그간의 고민들이 연상되었다. 음악 속에 논리가 있어야 하는 것은 당연하지만 그 논리들을 엮는 것은 작곡가의 직관이 결정한다는 것이다. 어쿠스틱적인 면이 강하며, 음향적인 요소보다 음악적인 진행에 역점을 두었다고 느껴진다.

페스티벌의 종반부로 다가갈수록 소리는 더욱 더 미래적이고 우주적으로 향해갔다. 네 번째 작품은 독일 작곡가 올리버 슈넬러Oliver Schneller의 바이올린, 비올라, 첼로, 일렉트로닉스를 위한 《스트링 스페이스String Space》이었다. 작품의 착상은 상상의 현악기가 공간에 매달려 있는데 그 현악기는 감각의 차원을 넘어서 있으며 이 거대한 현악기에는 바이올린, 비올라, 첼로가 걸려 있다. 모든 것은 상상 속에 존재하며 절대로 가능하지 않은 시나리오에 의해 이들의 관계가 형성된다. 진동, 떨림, 리듬, 공명 등은 그 '관계'에서 결정된다. 상대성 이론을 연상케 했다.

다섯 번째 작품은 편의상 여섯 번째 작품과 순서를 바꾸어 연주되었다. 아마도 컴퓨터 프로그램 사용의 편의를 위한 변화라 보인다. 컴퓨터음악 공연에서는 흔히 발생하는 일이라고 한다. 이돈응의 바이올린, 비올라, 첼로, 일렉트로닉스를 위한 《소리 추적 II Sound Trail II》이었다. 과거의 소리는 현재의 소리와 모종의 관계를 맺으며 미래의 소리를 만들어 나간다는 것이 주제이다. 이미 들려진 과거의 소리는 그 자체가 피드백을 거치며 확장되는가 하면 현재의 음들에 의해 변조되어 현재의 음과 함께 다시 들려진다. 제목 그대로 소리의 경로를 따라가며 '변하지 않는 것은 없다'라는 것을 새삼 깨닫게 한다.

페스티벌의 마지막 작품은 독일 안드레 바르테츠키Andre Bartetzki의 피아노, 타악기, 현악 3중주, 일렉트로닉스와 프로젝션을 위한 《인사이드아웃Insideout》이었다. 주제는 뫼비우스 띠를 바라보는 작가의 다양한 관점의 변화이다. 안과 밖의 구별이 없는 도형, 실제상과 거울상이 겹치지 않는 구조, 뫼비우스는 육안으로 보면서도 믿기 힘든 형태임에 틀림없다. 작가는 뫼비우스의 띠를 연상케 하는 다양한 형태들의 조합과 변종을 스크린에서 보여줌과 동시에 이에 상응하는 소리를 들려준다. 작가는 기하학의 논리를 이해하고 육안으로 체험하면서 느낀 경이로움을 작품으로 옮긴 것이다.

맺으며

추석연휴를 지내자 마자 시작했음에도 불구하고 컴퓨터음악제2015는 성공적으로 개최되었다. 청중의 대부분이 컴퓨터음악 관련 종사자 및 마니아로 구성되었다고 보이며, 작품에 대한 반응은 비교적 호의적이었다. 최첨단 디지털시대, IT 강국 대한민국에 살고 있지만 컴퓨터음악, 특히 동시대의 컴퓨터음악 분야는 '디지털'임에도 불구하고, '현대'적임에도 불구하고 대중적인 관심과는 다소 거리가 있었다. 그러나 컴퓨터, 현대음악이라는 정체성을 명확하게 표방해 왔기에 컴퓨터음악제가 20년이 넘는 시간 동안 국제페스티벌의 면모를 확고히 갖춰 나가며 성장할 수 있었다고 생각한다. 시사성이 강한 작품 및 소리 자체에 몰입한 작품, 음향효과에 역점을 둔 작품을 비롯하여 철저히 음악적 논리에 관심을 집중시킨 작품에 이르기까지 다양한 사유들이 구체화되었다. 개인적으로, 매년 개최되는 페스티벌인 만큼 해마다 소주제를 달아서 역사적인 흐름을 만들거나 페스티벌 자체의 슬로건을 만들어 개별화시키는 것은 어떨지 상상해 보았으며, 컴퓨터 음악제를 찾는 커뮤니티의 성격을 면밀히 파악함으로써 교육적이면서도 문화교류적인 역할을 확장시키는 것도 좋으리라 생각해 보았다.

CALL FOR WORKS / Seoul International Computer Music Festival 2016

The Korean Electro-Acoustic Music Society is proud to announce the Seoul International Computer Music Festival (SICMF) 2016. SICMF 2016 will be held from Óctober 14 to October 16 at Jayu Theater in Seoul Arts 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 2012.
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 5.
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)

Tuesday, 2 February 2016, 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 2016 작품 공모

한국전자음악협회는 서울국제컴퓨터음악제 2016에 연주될 작품들을 공모합니다.
서울국제컴퓨터음악제 2016는 10월 14일(금)부터 10월 16일(일)까지 열릴 예정입니다.

공모 분야

1. 테이프 음악
2. 악기(5명이내)와 전자음악 (테이프 혹은 라이브)
3. 라이브 전자음악
4. 오디오-비주얼 미디어 작품

공모 규정

1. 작품은 2012년 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)

2016년 2월 2일 (화) 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 방문하는 경우, 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 2016 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 2016는 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

2016년 3월 7일(월) 오후 4시 (당일 우체국 소인 우편물은 유효합니다.)

보낼 곳

서울특별시 성동구 행당동 17번지 한양대학교 1층 전자음악조교실 Tel. 02-2220-1709

응모 규정 및 참고 사항

1. 1981년 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 21일(월) 한국전자음악협회 홈페이지(http://www.keams.org)에 공지됩니다.
6. 공연당일 실연 심사를 통해 공연의 최우수작은 '서울국제컴퓨터음악제 2016'에 초대될 수 있습니다.

더 자세한 문의 master@keams.org

The Korean Electro-Acoustic Music Society's 2016 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 2016 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 2016: http://www.computermusic.or.kr/) from October 14 to October 16, 2016. Selected papers from the conference will be published in *Emille* Vol. 13 (December, 2016). If you want your paper or workshop idea to be considered for the 2016 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 5, 2016 (Korean Standard Time, UTC+9)
- Notification of Acceptance of the Proposal:	March 1, 2016 (Korean Standard Time, UTC+9)
- Deadline for Paper Submission before Conference:	August 31, 2016 (Korean Standard Time, UTC+9)
- Conference:	October 14-16, 2016 (Korean Standard Time, UTC+9)
- Notification of Selected Paper for the Journal Emille:	October 25, 2016 (Korean Standard Time, UTC+9)
- Deadline for Final Paper Submission:	November 10, 2016 (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.

2016년 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. 2016년 학술대회는 10월 14일부터 10월 16일에 열릴 2016 서울국제컴퓨터음악제(Seoul International Computer Music Festival 2016, SICMF 2016: http://www.computermusic.or.kr/)와 때를 같이 하여 열릴 예정이며, 이때 좋은 반응을 얻은 연구 결과들을 중심으로 2016년 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:	2016년 2월 5일
- 계획안 승인 통보일:	2016년 3월 1일
- 학술대회 이전 원고 제출 마감일:	2016년 8월 31일
- 학술대회:	2016년 10월 14-16일
- 논문 게재 여부 통보일:	2016년 10월 25일
- 최종 원고 마감일:	2016년 11월 10일

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.