array

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the journal of the ICMA

array

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Farewell

by Tom Erbe

Dear ICMA members,

I just wanted to write a short letter to announce my resignation as the President of the ICMA, and to welcome my replacement, Dr. Richard Dudas.

Richard has been a member of the ICMA for 20 years, and has served on the ICMA board from 2008 to 2015. He is knowledgeable and has played a part in all of the ICMA activities, and has shown steadfast dedication to our organization. After some discussion, the ICMA board voted unanimously (10 - 0) to approve Richard's appointment as president.

Myself, I will continue as an at-large board member for one more year, and will be assisting Richard in any way needed.

My very best,

Tom Erbe
UC San Diego Computer Music

Letter from the President

by Richard Dudas

What a fantastic start to 2017! I am both humbled and honored to have been elected to serve as President of the International Computer Music Association, after having served two terms as at-large board member. A big thank-you to ICMA members and board of directors alike.

My involvement with the ICMA has been longstanding: I first attended the International Computer Music Conference two decades ago – the 1997 conference in Thessaloniki. The experience was eye-opening both technically and musically, and I was able to finally meet many of the people who I had previously only known by either name or reputation. That year, I also wrote my first concert review for this very publication – the ICMA's journal Array. Since that first conference experience, I've been regularly involved with the ICMC in a multitude of capacities, attending the conference as both author of papers and composer of music, volunteering my time behind the scenes as a reviewer for the paper and music selection committees, and taking on the role of Paper Chair for the organizing committee of the 2015 ICMC at the University of North Texas.

It goes without saying that the experiences and opportunities arising from the ICMC would not be possible without the dedicated ICMA board members, listed at the front of this publication, who volunteer their time throughout the year to help the conference organizers put on a great conference. Among these, I'd like to take a moment to thank our outgoing president, Tom Erbe, who has helped strengthen and streamline the ICMA over the past several years, and who will continue to serve as at-large board member. Furthermore, since the previous issue of Array, there have been several new faces elected to the ICMA board whom I'd to welcome: Miriam Akkerman, John Thompson, and Mark Ballora were recently elected at-large board members, and Charles Nichols and Takeyoshi Mori as Regional Directors for the Americas and Asia/Oceania, respectively. I am looking forward to collaborating with them, alongside the seasoned board members, to continue to improve and nurture the ICMA and its community.

It is with sadness that I mention the loss of some pioneers and influential members of our electronic and computer music community: Pauline Oliveros, Jean-Claude Risset and Pierre Boulez. I hope the short memorials to them published in this edition of Array will serve as solicitations for more extended memories and anecdotes from you – the members of the ICMA – to be published in the next issue.

And, as always, being committed to our primary purpose – putting on and improving the annual ICMC – we are looking for potential hosts for future conferences. If you have been considering hosting an ICMC, please contact the Vice President for Conferences, Meg Schedel, or myself, so we can "get the show on the road"!

The greater computer music community's steadfast creativity and its adeptness in sharing ideas for the common good are hopefully also reflected in the ICMA. The association and its eminent conference have provided marvelous opportunities for developing and established computer musicians alike to be awestruck and enlightened, as well as to meet, exchange thoughts, and remain connected over large distances. It will be a privilege, as president, to be able to help the ICMA continue to provide current members with the kinds of memorable experiences and exchanges it has provided me, as a member, over the past decades.

Looking forward to meeting many of you at the 2017 ICMC in Shanghai!

Richard Dudas
President, ICMA

Letter from the Editor

by Christopher Haworth

Welcome to the 2016-17 issue of Array, the Journal of the International Computer Music Association. The focus of this issue is two keynote speeches from ICMC 2015 that give wildly different perspectives on the theory and practice of electroacoustic and computer music: Miller Puckette's 'The Sampling Theorem and its Discontents', and Jonty Harrison's 'State of the Art? A personal reflection on the intersections of music, sound and the creative imagination'. I am also pleased to publish a featured article by Elizabeth Hoffman, 'What is at Stake in the Politics of Digital Music Archive Access Policies? A Brief Look at Some Evolving Issues'. This fascinating article takes a comparative look at some emerging paradigms in digital music archiving, highlighting key challenges facing composers and musicologists as the materials of music migrate online. In the reviews section, Laurie Radford takes a long and informed look at Peter Elsea's The Art and Technique of Electroacoustic Music, while Lauren Hayes and Jonathan Higgins review concerts from ICMC 2016. Finally, following the sad news that three computer music pioneers died in 2016, I am pleased to be able to bring you thoughtful reflections on the lives and musical contributions of Pauline Oliveros by Meg Schedel; Pierre Boulez by Arshia Cont; and Jean-Claude Risset by Chryssie Nanou.

The next issue of Array will be devoted to showcasing the work of women and other minority groups in computer music. Guest edited by Shelly Knotts and Patricia Alessandrini, and drawn from a Women in Sound / Women on Sound event held earlier in the year, it is hoped the issue will build on the work being done to diversify the field and enhance the visibility of non-hegemonic groups.

I am always happy to receive suggestions for featured articles, requests to review books, CDs, or concerts for Array, letters in response to articles, or related things. Please send these to christopher.p.haworth@gmail.com

Pauline Oliveros, 1932-2016

by Margaret Schedel

When I heard the news that Pauline Oliveros had died on Thanksgiving Day I was completely gutted. I mourned of course for her partner Ione, her long-time collaborator Heloise Gold, and all of the members of the Deep Listening Community who have become my extended family. Pauline was an incredible performer and composer, but she went beyond the boundaries of a traditional musician and created a listening practice that has formed multi-generational friendships among computer musicians, acoustic composers, and musical improvisers (Pauline once famously called for 'improvisatories' in addition to conservatories). Even though many of us haven't met in person, we form a vibrating bond across the planet that never ceases to amaze me. Deep Listening Retreats have almost become a rite of passage for women in experimental music, and a safe space for all to listen, to sound, to move, and to dream. I first attended the retreat under the encouragement of my own composition teacher, Mara Helmuth, who had always found them so inspiring. I met my long-time collaborator (and bridesmaid) Sarah O'Halloran at one retreat, and formed other new and deep friendships in those open and daring weeks we spent together.

It took me a while to figure out why I was so gutted by the news. For a while I assumed it was because it was so unexpected. Pauline's health was not in decline, and her own mother had lived well into her nineties; furthermore, I had seen her fairly recently and she was as vibrant as ever. After a meditation I finally realized why I was so sad. For the first time, I had students of my own who I wanted to encourage to attend a retreat – three people I thought would benefit from her wisdom and abilities as a composer, performer and educator. I was looking forward to introducing them to her; to seeing her simultaneously listening to them and challenging them; to seeing the awe in their faces turn quickly to companionability, as Pauline made them comfortable. I was sad because I would never be able to participate in a sonic meditation with them, led by her unwavering energy as she improvised complex sounds from simple rules. They were never going to hear her belly laugh, or see her grounded and uplifting performances – this is what made me sad. I can only hope that her spirit will continue to inspire, as it already is doing with the spontaneously organized tributes that recognise her tremendous impact.

I first met Pauline at a workshop in a summer program at the Kitchen. Of course I knew of her, and her music, but I wasn't as familiar with her sonic meditations. We performed her piece, *Interdependence*. In it, every performer self-selects a role as 'sender' or 'receiver', and performers can change roles at will. In the first section, senders play one short note at any time. The note can have any pitch and dynamic, but must be played with 'intention'. Receivers then play a short note as fast as possible in response – also with any pitch and dynamic. In the second section this changes: receivers are allowed to sing long notes in response, rather than short ones; and in the third section, it is the senders that can sing long notes; receivers are instructed to react as quickly as possible to the end of the note. The score reads:

> The correct player reactions can create an atmosphere of electricity that runs through the ensemble in a rippling effect. These ripples of pitches will be in random patterns depending on the decisions of the players. A ripple could be short (one sender with two or three receivers) or longer depending on the decisions and reaction times of the players. An effective reaction time means that the player is aware of their own response slightly after the reaction has already happened (milliseconds). The variations introduce long tones which develop into chords and textures inside of the ripples [1].

Our group stayed within the confines of the instructions for a while, but then started to experiment with glissandi, short phrases, and percussive effects. Pauline let the piece come to a natural close, and then told us that she was very aware we had gone beyond her instructions. Indeed, she declared that only the first part of the improvisation was her piece – she could be very strict! She then took out her accordion and proceeded to replay our entire performance from memory, winking as she transitioned into our free improv. She captured the nuance of our group sound and echoed it back to us, shimmering with precision. Pauline had an astonishing ear for music, and delighted in trying new things. This was evidenced by her constantly evolving compositions and performance practice.

Pauline was not just a composer and performer – she was also a community builder. As I write this, immigrants, refugees, and temporary and permanent residents are being detained at the United States borders. All three people I wanted to meet Pauline have been personally and directly impacted by the actions of the incoming Trump administration. Learning from my lessons with Pauline at the retreats, I listened deeply to their concerns and held space for them. We are now taking action. Even though they never got the chance to meet Pauline, in my mind they have become part of the Deep Listening Community, and I know others will continue to widen the circle as well.

Pierre Boulez, 1925-2016

by Arshia Cont

In 2016 the music community lost many important figures who left their mark on the 20th century, and the Computer Music community has been no exception. Among such figures, Pierre Boulez is one whose loss will be mourned by the classical music, contemporary classical music, as well as the computer music communities. Whereas his global impact as a dominant figure (both composer and conductor) of the classical music world is widely recognized; his legacy, impact and longtime involvement in the computer music community deserves further attention.

Still in his 20s, and after an early and controversial career in avant-garde instrumental music, Boulez entered the Musique Concrète group led by Pierre Schaeffer in the late 1940s. In addition to composing a few Concrète pieces in this period, Boulez was the person playing some of the piano sounds in Schaeffer's *Cinq études de bruits*. His experience with Musique Concrète, followed by the early foundations of the Darmstadt School with fellow composers Luciano Berio, Luigi Nono and Karlheinz Stockhausen, convinced the young Boulez that the renewal of material alone is not enough for an intellectual restructuring of music history. He did not hesitate at that time to compare the evolution of musical materials to that of architecture and construction. He further believed and acted upon, that such changes cannot happen by a single person's effort but by coordinated efforts in science, art, society and politics.

In the 1970s, at the height of his artistic career as head of the New York Philharmonic and BBC Symphony Orchestras, he was called by the then French president Georges Pompidou to create a world-class orchestra in Paris. Boulez responded with an alternative proposal to create a center where scientists, musicians and artists could work together; he took the Bauhaus as his model. This led to the creation of IRCAM in 1977, which in its early days brought figures who would go on to become key architects of electroacoustic and computer music – Jean-Claude Risset, Karlheinz Stockhausen, Luciano Berio, Max Mathews, and David Wessel – together to build a utopia at the intersection of science and music, but focused on musical creativity. IRCAM would become home to software and technology developments that would become central to the computer music communities, going far beyond the musical work of Boulez himself (and Max/MSP is but one example). As a conductor and musical director after he left IRCAM in 1992, Boulez was a strong supporter of young computer music composers, inviting them to festivals and orchestras all around the world and initiating the necessary infrastructure to support their creativity.

The imprint of Pierre Boulez' activity is visible not only in his acclaimed compositions and writings, but also in his support and influence on composers, technologists and researchers of our community who continue to channel his force, intellect and generosity.

Jean-Claude Risset, 1938-2016

by Chryssie Nanou

At the end of the summer of 2014 we gathered on the Hill of the Muses, overlooking the Acropolis, to hear the music of our contemporaries performed under a clear Athenian night sky. As part of the joint SMS/ICMC held in Athens, this wonderful evening concert featured works diffused live in a beautiful – but noisy – setting, surrounded by aged pines, Hellenic crickets and the noise cloud of a bustling city.

It was at the end of the program when our dear friend Jean-Claude Risset took his seat smiling to diffuse the four-channel work *Elementa* (1988), a four-movement, 22 minute tour de force compositional mastery and ingenuity. At the core of his compositional thinking – mixing natural with synthetic sounds, merging vocabulary and syntax – is a captivating voyage of a living organic soundscape. Much to our surprise the piece, instead of losing definition and precision in the wash of background noise, shone through in the natural environment creating an amalgam where the composer's personality, the "man of the South", was revealed. It was Jean-Claude in his element, one of our most sophisticated thinkers and skilled artisans mixing a captivating and enrapturing piece within the natural environment of the Mediterranean South.

For all of his pioneering work around the globe, Jean-Claude was infinitely generous with his time and spirit. He guided and encouraged generations of musicians and scientists alike, from Bell Labs to IRCAM, CCRMA to Dartmouth. When my friends and I share stories of wonderful times spent with Jean-Claude, they ultimately end in a similar discussion of his generosity of time, spirit and wisdom.

Jean-Claude Risset passed away in Marseille on November 21, 2016. On behalf of the ICMA Board of Directors and the membership as a whole we thank Jean-Claude for his vast scientific and musical contributions to the field, as well as his tireless energy and enthusiasm for art and for life.

References

[1] Oliveros, Pauline. 'Interdependence', In: *Four meditations: for orchestra*. Deep Listening Publications, 1996.

ICMC 2015 Keynote Address The Sampling Theorem and its Discontents

by Miller Puckette
Saturday, 26 September 2015

The fundamental principle of computer music is usually taken to be the Nyquist-Shannon sampling theorem, which states that a band-limited function can be exactly represented by sampling it at regular intervals. This paper will not quarrel with the theorem itself, but rather will test the assumptions under which it is commonly applied, and endeavor to show that there are interesting approaches to computer music that lie outside the framework of the sampling theorem.

As we will see in Section 3, sampling violations are ubiquitous in everyday electronic music practice. The severity of these violations can usually be mitigated either through various engineering practices and/or careful critical listening. But their existence gives the lie to the popular understanding of digital audio practice as being 'lossless'.

This is not to deny the power of modern digital signal processing theory and its applications, but rather to claim that its underlying assumption – that the sampled signals on which we are operating are to be thought of as exactly representing band-limited continuous-time functions – sheds light on certain digital operations (notably time-invariant filtering) but not so aptly on others, such as classical synthesizer waveform generation.

Digital audio practitioners cannot escape the necessity of representing continuous-time signals with finite-sized data structures. But the blanket assumption that such signals can only be represented via the sampling theorem can be unnecessarily limiting. In Sections 4 and 6 I'll describe investigations by two recent UCSD graduates that each adopt a distinct approach to audio manipulation outside the framework of the sampling theorem.

A collection of accompanying patches that demonstrate some of these ideas can be downloaded from msp.ucsd.edu/ideas/icmc15-examples/.

1. The assumptions

Band-limited functions are a vector space: you can scale one of them, or add two of them, to get another. But that is where closure ends. The trouble begins as soon as we even go so far as to multiply one signal by another. Suppose two sampled signals, $X[n]$ and $Y[n]$, are used to represent two continuous functions of time x(t), y(t), which we assume to be band-limited, containing only frequencies in the Nyquist frequency band, the interval $(-R/2, R/2)$ where R is the sample rate.The values can either be real or complex, and for simplicity we'll assume the computer can exactly represent the numerical values. (It isn't true but that is usually a comparatively minor issue).

There is, of course, a perfectly good continuous-time signal, call it $z(t)$, that is represented by the computable product, $Z[n] = X[n]Y[n]$. But it's not in general the case that $z(t) = x(t)y(t)$. We didn't in reality make the product of the two continuous-time signals we were representing when we multiplied their computer representations.

At this point we can look ruefully back at every occurrence of the character "*" in all the Csound, Pd, SuperCollider, Kyma, 4X, or MUSIC 10 instruments we've ever built and reflect on the fact that the result isn't really correct, if we regard our sampled signals as representing continuous-time ones. Often it's a very serviceable approximation. If, for instance, the signals $x(t)$ and $y(t)$ have frequency limits whose sum is less than $R/2$,the multiplication is exact; and when not exact, it is often a very good approximation. But the approximation's accuracy or lack thereof is rarely worked out explicitly.

We could always take action to band-limit two signals (by filtering them) before multiplying so that the multiplication itself doesn't yield frequencies outside the Nyquist frequency band. But this would cause delays and/or phase distortion, not to mention the computational cost this would incur.

One fundamental operation in electronic music practice (in my thinking, the most fundamental one) is table lookup, which is used in digital oscillators and samplers, and also in nonlinear techniques such as FM and waveshaping. Again sidestepping the comparatively minor issue of the accuracy limits of wavetable lookup, we instead again consider the possibility of frequency products landing outside the Nyquist band. Suppose the incoming signal is a sinusoid of frequency ω and that the wavetable lookup can be approximated as a power series,

$$f(x) = a_0 + a_1 x + a_2 x^2 + \ldots$$

The highest possible frequency product of the kth term $(a_k x^k)$ is $k\omega$. If the function is a polynomial (thus stopping at a finite k)

then the situation is at least in principle possible to control by limiting *kω* never to exceed *R/2* (whether by fiat or by filtering). But for any other choice of *f(x)* the result is in general not band limited at all, and some foldover is inevitable.

There is a reasonably broad class of operations that can be carried out without departing from the safe zone of the Nyquist theorem. One can record and play back sounds. Delay networks and filters (both recursive and not) are safe as long as the coefficients do not change in time. One can spatialize sounds using level panning. But this still leaves a majority of electronic music practices that cannot be guaranteed band-limited in practice; in addition to the examples of FM and waveshaping cited earlier, even additive synthesis, which would seem to be safely band-limited at first thought, is in reality not, since at a very minimum we have to multiply the component sinusoids by time-varying envelopes.

If, for example, these envelopes are constructed using line segments, then every envelope breakpoint gives rise to non-band-limited frequency products dropping off in amplitude as ω^{-2}. The resulting foldover is often inaudible but it is not hard to concoct situations in which it is not.

The most ready defense against the distortions arising from digital sampling is to train one's ears to hear it and, as necessary, adjust parameters or raise sample rates until it is no longer audible But to learn to hear this, a young electronic musician would need examples of clean and dirty signals to compare. It is possible that practice will erode over the years as ears gradually get used to hearing foldover, until perhaps one day few people will have heard a cleanly synthesized sound, in much the same way that few North Americans or Europeans have ever tasted a tree-ripened banana.

2. Example of a non-band-limited signal representation

Any system for representing continuous functions digitally will only be able to exactly represent a small subset of all possible functions, and/or to approximate, more or less well, functions that can't be exactly represented. Any particular choice of representation will imply a certain subset of functions that can be represented, and perhaps a way of choosing which representable function to swap in for one that is not representable. For example, sampling at a constant rate allows us to claim the subset of functions that are suitably band-limited and to approximate any other one by leaving out whatever lies outside the band limit. This is clearly an excellent choice for digital audio in general, but for some applications other choices might be preferable.

Here for example is another possible choice: we could choose to represent arbitrary piecewise linear functions of time by specifying the endpoints of the line segments. For example, a function like the one shown in Figure 1 could be represented by a sequence of triples:

$(t_1, x_1, y_1), (t_2, x_2, y_2)$

This would allow, for example, a sawtooth wave to be represented exactly. Certain operations (adding two such functions together, for example) could be carried out in the representation, but others (for instance, multiplying them) could not (although we could allow that as well if we extended the format to allow arbitrary piecewise polynomials... but I won't belabor the point here).

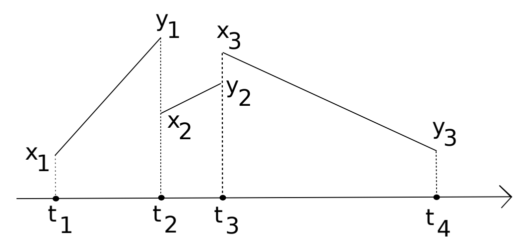

Figure 1: A digitizable representation of piecewise-linear functions of time

The interesting thing about this format is that it can exactly represent classes of functions that can't be represented using the sampling theorem. Although it is certainly less well adapted to the day-to-day operations of most electronic musicians than sampled functions would be, there is at least one piece of music that would have been quite naturally expressed in this way: Xenakis's *S709*, a few microseconds of which are shown in Figure 2, and which is described in [9] with an appendix showing a code listing of Marie-Helene Serra's implementation. The piece is realized by generating repeated copies of a line-segment waveform in which the vertices vary at random, successively from cycle to cycle; the number of segments may vary as well. This is at least an existence proof that a line-segment-based signal representation may lead naturally to signal manipulations that at least some composers might find musically useful.

3. Violating the theorem's assumptions

On the subject on the sampling theorem, we should not forget that the whole practice of electronic music using sampled audio signals, and indeed the now-ubiquitous use of wavetables for sound synthesis, dates back to Max Mathews. Mathews himself was trained as an engineer and always took care to let people know about the limitations of the technology. Around 2007 he was showing visitors to his laboratory a wonderful demonstration which, since I haven't seen it published, I'll repeat here.[1]

Mathews's idea is to put a square pulse in a wavetable (in my example, I put a one-sample-wide pulse in a 200-element table) and then to scan it, without interpolating, with the phase advancing by various sampling increments. Choosing a sampling increment of *1/n* where *n* is an integer (taking the table lookup domain to be from 0 to 1), you get a clean pitch of *R/n* where *R* is the sample rate. (This assumes that the phase accumulation itself is done to arbitrarily high precision before applying the non-interpolated table lookup). Choosing an arbitrary sample increment gives a characteristically dirty sound.

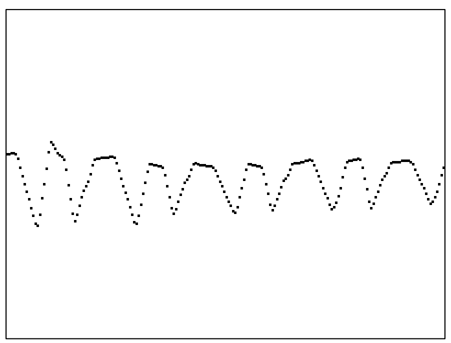

Figure 2: 200 samples of Xenakis's *S709*

We now choose a sample increment almost equal to 1/100 but slightly detuned. If the sample increment were exactly 1/100, you would either hear a sound if the phase happened to pass between 0 and 1/200, but silence if the phase passes between 1/200 and 1/100 but silence if (thereby skipping over the pulse). Since the slight detuning makes the phase drift alternately between these two cases, we unexpectedly hear a tone that toggles on and off. You can think of this as a beating pattern between an infinite series of foldover products that just happen to line up to make a square-wave modulation of the tone.

4. Example: modeling the Moog ladder filter

We now consider one interesting way to approximate continuous-time processes in a computer, using numerical differential equation solvers instead of sampled processes. In this discussion I'll rely heavily on work by recent UCSD PhD graduates Andrew Allen [1] and David Medine [5].

A good starting example is the famous Moog ladder filter design [6], a conceptual block diagram view of which is shown in Figure 3. Each low-pass filter in the diagram is a one-pole design whose cutoff frequency (k, in radians per unit time) is voltage-controlled. (Here we're not showing the elegant circuit design that realizes this; Moog not only had to find a good signal processing model but also one he could realize with bipolar transistors).

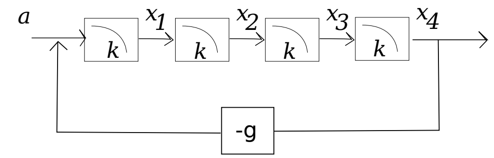

Figure 3: block diagram representation of the Moog ladder filter

This block diagram leads to the following system of ordinary differential equations:

$$x_1 = k \cdot [S(a - gx_4) - S(x_1)]$$
$$x_2 = k \cdot [S(x_1) - S(x_2)]$$
$$x_3 = k \cdot [S(x_2) - S(x_3)]$$
$$x_4 = k \cdot [S(x_3) - S(x_4)]$$

Here *S* denotes a nonlinear saturation function reflecting the fact that in any real circuit realization of the network, the filters' output would be limited by the available power supply. This is a good thing of course, because the filter can be made unstable by turning up the feedback gain *g*. We'd rather allow the outputs of the filters to saturate than merely vaporize the planet as would otherwise happen when *g* first exceeded 4.

The usual and somewhat schematic explanation of how this filter works is that, at the frequency *k*, each low-pass filter retards the signal by 1/8 cycle, so that the four of them retard it by 1/2 cycle, so that multiplied by *-g* the feedback path is in phase with the input (at *g/4* times the amplitude), so that the circuit resonates. The difficulty of digitizing this circuit stems from the fact that in a digital realization there will be at least a one-sample delay in the feedback path, thus changing the frequency at which resonance occurs. This change can be quite significant; for instance, if *k* is set to one quarter of the sample rate we pick up a fifth quarter-cycle, so we would expect the resonant frequency to be off by a minor third (20%). The filter is often used as an oscillator, in which usage this will be heard as a tuning error – and it would be reasonable to ask that one control an oscillator's frequency to within a few cents, perhaps 1000 times better than the naive digital implementation does. If we assume linearity this can be corrected satisfactorily using standard DSP techniques [7]; but if we take the nonlinearities fully into account it takes much hard work [3] to overcome the problems that result from digitizing the Moog ladder filter.

What I propose here will sound facile, and perhaps it is: why not go back to the differential equations and apply a traditional numerical ODE solver to them? Very little brainpower is required. One simply goes to the Wikipedia page for "Runge-Kutta" and types the familiar four-step version into a Pd extern. This is the basis of the bob~ object released with Pure Data.

This approach has the disadvantage that it requires far more computation to generate output samples than the DSP approach does. If your end goal is a stand-alone software or hardware product that imitates the historical Moog ladder filter, it may well be worth the research and development time (months or years) required to implement one using the work cited above. But on the other hand, if your aim is to explore one or another possible refinement of, or deviation from, the modeled filter then you would have to redo all this work for each possible modification. Furthermore, without any real filter to test your results against, you could never know how accurate your modeling really is.

For one thing, we cannot automatically assume that the many idealizations built into our model aren't causing us to lose something in translation [8]. To know that for sure we would have to make comparisons, one by one, of the simplified model against one in which each simplifying assumption was replaced with a more realistic one. This is feasible using numerical methods, but would be onerous to do using DSP techniques.

But things get even more interesting when we consider possible variations on the filter design itself (leaving aside the question of whether a 'real' circuit might exist to exhibit them). After all, there is something self-defeating in the idea of using contemporary technology to try to recreate sonic experiences from the past, when instead we could be looking for new ones.

To make just one example, suppose we decided that the cutoff/resonant frequency k should depend on the internal state of the filter, for instance taking one value when state variable x_1 is positive and a different one otherwise. If you drove such a filter to oscillation ($g \geq 4$) you would get a sort of self-FM, and if instead (or in addition) you drove it with an incoming sound you could get a variety of effects.[1]

You could make all sorts of other changes; for instance changing the number of stages from four to eight or twelve, possibly making several taps with independently controllable feedback coefficients, inserting input signals at more than one point in the circuit, making the saturation function asymmetrical, and so on without end.

This line of exploration should not be confused with the idea of simulating or modeling actual circuits. One could do that with the SPICE circuit simulator, for example. But such an approach has several disadvantages. First, you have to design a real circuit, which is much harder to do than to arrange low-pass filters as described in the functional not yield explicit expressions for the derivatives of the state variables; instead, a system of simultaneous equations must be solved to compute the derivatives. (In physical terms, this is because real electronic components don't have 'inputs' and 'outputs'; instead, causality flows bidirectionally along each physical wire.)

Instead, what we have here, as David Medine proposes, is a block-diagram-based system of components each of whose output's derivative is a function of its state and inputs, in such a way that we can construct a modular synthesis environment that is realizable in systems of differential equations in explicit form, readily solvable using straightforward techniques such as Runge-Kutta. Although the software doesn't exist yet, this could easily be made into a graphical patching language for quickly exploring a wide range of synthesis techniques.

5. Two More Dynamical Systems

The Moog filter simulation above is an example of a dynamical system, which is only to say, 'it's a system that can be written as a set of simultaneous first-order differential equations, solved for the derivative terms'. Such a system can be visualized as in Figure 4.

Here the system of equations describes a simple forced oscillator:

$$\dot{x} = -ky + (1 - x^2 - y^2)\, x + f(t)$$
$$\dot{y} = -kx + (1 - x^2 - y^2) y$$

This can be thought of as a vector field, where the points are possible states of the system and the vectors are the time derivatives which show how the current state flows through the state space. The flow may depend on time; in this example there's a forcing function $f(t)$ imposed from elsewhere. (The vector field is drawn in the figure with the forcing function $f(t)=0$).

When $f(t) = 0$, this oscillator converges to the unit circle where the term $1 - x^2 - y^2$ disappears; the result is simple harmonic motion. As with the Moog filter when pushed into oscillation, this example gives various results when forced with a sinusoid tuned a minor third or so from the natural oscillating frequency.[1] (In truth it is much less interesting sonically than the Moog example, but its conceptual simplicity makes it suitable for a range of extensions that will not be explored here.)

For another example of a dynamic system, consider the famous Lorenz attractor.[1] Here, for convenience, in addition to the usual parameters α, β, ρ there is a speed parameter, in MIDI units, that simply scales all the time derivatives so that the model runs globally faster or slower. The output can either be listened to directly (by connecting one or another state variable directly to a loudspeaker) or used to control the pitch of a sinusoidal oscillator; I find the latter choice the more interesting to hear.

This approach has the disadvantage that it requires far more computation to to directly (by connecting one or another state variable directly to a loudspeaker) or used to control the pitch of a sinusoidal oscillator; I find the latter choice the more interesting to hear.

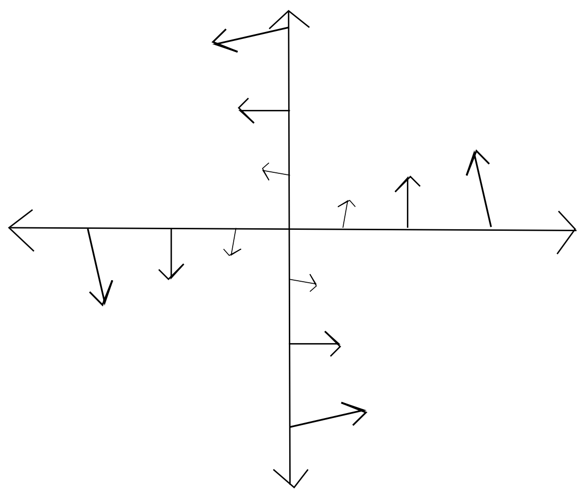

Figure 4: Example of a dynamical system: a forced oscillator

state variable directly to a loudspeaker) or used to control the pitch of a sinusoidal oscillator; I find the latter choice the more interesting to hear.

6. Ruratae: unphysical modeling

Andrew Allen takes continuous-time modeling in a quite different direction, realized in his Windows game (to use the word loosely), named *Ruratae*. Here the model is that of a physically vibrating network of interconnected objects, much as in physical modeling systems such as Cordis Anima [2] or Modalys [4]. Unlike those systems, the emphasis here is not on exactly modeling a real physical system. Such modeling has limitations similar to those of circuit modelers as either would be applied to music synthesis: expertise is required to 'build' reasonable sounding instruments, and once the instruments are built they cannot be quickly modified.

Ruratae takes a higher-level approach, in which fanciful collections of point masses are connected by generalized 'springs' that may exhibit nonlinear responses, damping, and/or may snap when elongated past a maximum value. The system makes no distinction between the act of building an instrument and that of playing it. The user hears the instrument vibrating in reaction as masses and connections are added or deleted (or snap). This encourages a highly intuitive and exploratory style of instrument design.

Compared to dynamical systems in general, *Ruratae's* focus on idealized physical systems constrains them in a way of the system in real time at computer-game-worthy frame rates (the graphical optimization was tricky and system-dependent, which is why the game runs only on Windows).

To draw a conclusion from the work of both Allen and Medine, the universe of ODE systems is still uneven terrain where no single approach is without its own particular set of limitations. At the same time, both approaches are powerful and offer much potential to build compelling and fun computer music instruments. This should continue to be an active area of research.

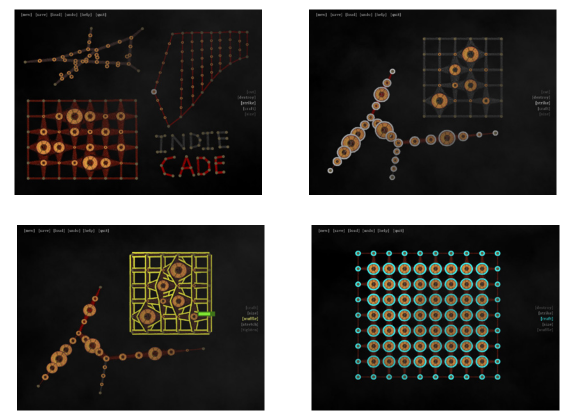

Figure 5: Screen shots from Andrew Allen's *Ruratae* software (reprinted from his PhD dissertation)

7 Uniform flows on locally flat surfaces

We turn now to a very different possible approach to modeling continuous-time processes. Returning to the idea of using dynamical systems as audio generators, we propose a methodology for designing ones for which we can find exact solutions despite the availability of interesting non-periodic behavior. Specifically, we can impose a constant vector field as the flow, so that locally we get motion in straight lines. Interesting results can come from connecting flat sheets together in geometries that have cantankerous global properties.

A physical system that suggested this approach is pictured in Figure 6. Two ideal mass-spring systems, with equal masses but tuned to different frequencies, are held at a distance apart so that they collide, either occasionally or constantly. Collisions are elastic: each mass recoils at its speed of incidence as if it had bounced to a hard surface. (This isn't really correct; the masses should in fact exchange velocities; but it is much easier to model this way since each oscillator's energy then stays fixed.)[1]

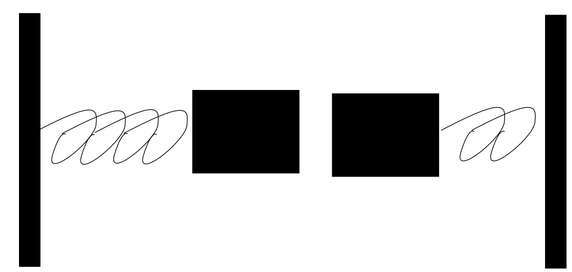

Figure 6: Dynamical system: two colliding,

Both of the two systems are assumed to oscillate with amplitude 1, and can thus be represented by their phases ϕ_1, ϕ_2, which we take to range from $-\pi$ to π, and equal to 0 when a spring is at its most stretched. At moments where the phases are such that the two masses come into contact, say at $\phi_1 = -\phi_a$ and $\phi_2 = -\phi_b$, we simply advance the phase so that they are in the same location but moving away from each other instead, that is, wrapping around forward to phases $\phi_1 = +\phi_a$ and $\phi_2 = +\phi_b$. To be exactly correct, we should measure by what amount the two phases have exceeded the values at which the collision occurs and the rebound phases should be forwarded by the same amount, but the provided patch does not take care of this detail.

Here is an analysis of the behavior of the system, slightly further simplified but presented in a way that can readily be generalized. The phase space is a square whose coordinates are the two phases, with a centered, diagonally oriented square corresponding to points at which the two masses would occupy the same space (see Figure 7. This is a simplification; in the original physical model the forbidden region is not really a square. Many other boundary shapes could be used instead.)

Without the middle square cut away, the phase space would be a torus and the flow would be a constant vector field, so that trajectories would be the familiar geodesics known to players of 1960s-vintage SPAWAR. The missing square acts as a wormhole in the space. Whereas the dotted path in the figure represents a possible trajectory in the absence of the wormhole (so that the two oscillators advance independently), in the presence of the wormhole the trajectory is altered as shown by the solid path.

We can then listen to any suitably smooth function of the phase space. For instance, to hear a mixture of the two oscillators we would choose the function $\cos(\phi_1) + \cos(\phi_2)$, but other choices abound. We would require only that the function take the same value on any two diametrically opposed points so that the result of crossing the wormhole is continuous. (If we wish, we could work somewhat harder and arrange for matching derivatives as well.)

The whole scheme could easily be extended to higher-dimensional spaces (representing more than two oscillators) with as yet unexplored results. Even with only two dimensions, a variety of rich interactions between the two oscillators can be quickly found.

The interesting thing about this model is that it allows for exact solutions. To know our position in phase space at any point

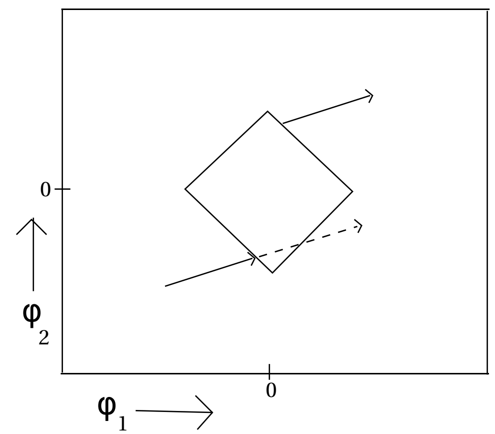

Figure 7: Trajectories through toroidal phase space: dotted path, normal; solid path: with wormhole

in time, we merely propagate forward in a straight line until we hit a boundary (at a time point that in general won't be an integer number of samples at any fixed sample rate). Whenever we reach a boundary, we jump to the diametrically opposed boundary point and continue as before. This gives us a list of segments in a format similar to that of Figure 1. To listen to the output, we convert it to a sampled signal.

8 Observations and conclusions

Early Bell-Labs-resident composers such as James Tenney, Jean-Claude Risset, and Charles Dodge set out a theory and praxis of computer music that many composers have since followed, privileging precise execution of carefully specified and planned-for musical desiderata. The hankering of late twentieth-century Western composers for order and structure fit in perfectly with the computer's ability to accurately manipulate data, and their musical practice did not suffer much from the computer's main early failing: the impossibility of real-time audio computations. It is in a spirit of appreciation for their contributions that I am here exploring the spaces beyond the pale they constructed – if for no other reason than the light it sheds on what we're doing as we follow in their footsteps.

Meanwhile, traditional musical instruments (especially that oldest one, the human voice) refuse to give up their secrets, and remain capable of musical gestures that no computer can yet imitate. Part of the secret undoubtedly lies in the real-time interaction between player and instrument, and perhaps another aspect is the complexity and inherent unpredictability of the physical processes that take place inside the instruments.

It is no accident that all the examples I have invoked here are in one way or another unpredictable. Because of this they practically require real-time exploration to unlock their musical possibilities. In this respect they are all also beholden to another tradition perhaps best exemplied by Michel Waisvisz's famous Crackle Box. They lie on the fringe of what is considered correct electronic music practice. Fringes are are interesting loci, and any reasonably complex domain will have many of them; so even if each individual one is limited in range their aggregate might offer a large range of possibilities. Besides, what seems like a fringe one day might be understood as the mainstream sometime in the future (for example: electronic music itself).

Notes

1.The examples used in this article are: mathews-table-lookupexample.pd, bentbob-test.pd, forcedosc-test.pd, lorenz-test.pd, coupled-sampled.pd. All are available from msp.ucsd.edu/ideas/icmc15-examples/

References

[1] A. S. Allen, "Ruratae: a physics-based audio engine," Ph.D. dissertation, University of California, San Diego, 2014.

[2] C. Cadoz, A. Luciani, and J. L. Florens, "Cordis-anima: a modeling and simulation system for sound and image synthesis: the general formalism," *Computer Music Journal*, vol. 17, no. 1, pp. 19-29, 1993.

[3] A. Huovilainen, "Non-linear digital implementation of the Moog ladder filter," In Proceedings of the International Conference on Digital Audio Effects (DAFx-04), 2004, pp. 61-64.

[4] F. Iovino, R. Causse, and R. Dudas, "Recent work around modalys and modal synthesis," in *Proceedings of the International Computer Music Conference*.

[5] D. Medine, "Dynamical systems for audio synthesis: Embracing nonlinearities and delay-free loops," *Applied Sciences,* vol. 6, no. 5, p. 134, 2016.

[6] R. A. Moog, "A voltage-controlled low-pass high-pass lter for audio signal processing," in *Audio Engineering Society Convention* 17. Audio Engineering Society, 1965.

[7] T. Stilson and J. Smith, "Analyzing the Moog VCF with considerations for digital implementation," in *Proceedings of the International Computer Music Conference*, International Computer Music Association, 1996.

[8] T. E. Stinchcombe, "Analysis of the Moog transistor ladder and derivative lters," Citeseer, Tech. Rep., 2008.

[9] I. Xenakis, *Formalized Music: Thought and Mathematics in Composition*. Pendragon Press, 1992, no. 6.

ICMC 2015 Keynote Address State of the Art? A personal reflection on the intersections of music, sound and the creative imagination

by Jonty Harrison
Sunday, 27 September 2015

I am delighted and honoured to be one of the keynote speakers for the 2015 ICMC, and I should like to take the opportunity, right away, to thank Jon and the organising committee for inviting me to take on this role. All the same, I'm a bit puzzled!

1. Why am I here?

As an ICMC keynote, I find myself in some impressive company. Recent keynotes, including my two colleagues at this conference, Carla Scaletti and Miller Puckette, have been true pioneers in the field of computer music composition and/or have developed programs or algorithms that have changed the shape of the computer music world. I have done neither of these things; I am a few years too young to belong to the 'pioneer generation', and my programming skills are embarrassing. So, without for a moment wishing to question Jon's judgement in inviting me, part of me nevertheless wonders exactly what I'm doing here today. Perhaps it was thought that my recently acquired emeritus status might give me some special insight on the conference's main theme – Looking Back, Looking Forward. (It is certainly true that my change of status has made me think about what I have done, do now and might do in the future.) Or, perhaps people thought I might spice things up a bit by being provocative. This is always possible – as I get older, I am increasingly curmudgeonly and critical of what I see (and hear) happening around me, and official retirement means I can say what I really think (what can they do – fire me?).

Whatever the reason for my presence, and despite my keynote address being scheduled between those of Carla and Miller, I hope that I can make a few observations about the state of the art – or, at the very least, the state of my art – that will make this presentation more than just a comedy interlude!

2. What do I do?

I shall assume, therefore, that I am standing here because of my work in the music part of the 'computer music' equation.[1] And, for the past 35 years, 'music' has meant three things for me: composition, performance and teaching. However, separating these three facets of my work is very difficult as, for most of what might loosely be termed my 'career', composition, performance and teaching have been inextricably intertwined. And this has had both positive and negative aspects, for there have been some years in which my total compositional output was 0'00" – thanks, in particular, to teaching (or, more accurately, to university administration, which seems the most time-consuming aspect of higher education these days).

3. My name is Jonty and I am… an acousmatic composer

Let's start with composition, as that is at the centre of my activities. Composition is what defines me – to myself. If I were not a composer, I would not be involved with performance and I would certainly have no justification for being in education. I am, first and foremost, a composer – an acousmatic composer, to be precise. But this was not always the case.
I am a classically trained musician: piano lessons at age six, horn player after that (not a bad one, actually, even making it into the National Youth Orchestra of Great Britain and I seriously considered trying to go professional), conductor, and member of a music theatre group at University (think Kagel, not musical theatre like Broadway and the West End of London). I am a composer who has always been involved in performance: an obvious but not unusual link. My musical passions during my teenage years were Wagner, Mahler, Debussy, Stravinsky and Schoenberg (though the Beatles, Bob Dylan and others also got a look-in!). My classical training has, of course, coloured my compositional thinking and has left audible traces in my music. For example, something that I can only describe as a sense of 'phrase' or 'phrasing' (even including a notion of 'cadence') when shaping musical time, as well as a related preoccupation with 'causality', have both found their way into my acousmatic music. I have an ingrained sense that, as a physical phenomenon, sound is related to and results from physical action. Sound does not just happen; it is made. As a consequence, my acousmatic music is articulated by gestural events, which appear to cause changes in the surrounding musical fabric. It may also explain the predominance in my work of 'real' sound materials over electronically generated ones – although there are plenty of those, too!

My age and the era in which I grew up are therefore important factors in my musical makeup. I am old enough to have been trained during a period – the 1960s and 70s – where composition was regarded as a highly intellectual activity, involving lots of pre-compositional pitch charts, durational schemes, and so on. At the University of York in the early 70s, discussion among composers (and there were very many of us, including all the faculty members!) revolved largely around what we would now see as the high modernist project related to integral serialism, and around its leading figures – Boulez, Stockhausen and Berio (with whom my tutor, Bernard Rands, had himself studied). I was a fully signed-up member of this club, and I must confess that I am still a huge fan of much of this music.

But then something strange happened. After four years at York, and at the end of my first year as a graduate student, I decided that I should find out what the electroacoustic music studio had to offer me as a composer. What I expected to find there were ways of extending and expanding what I was already doing in the instrumental domain. What I actually discovered challenged everything I believed about what music was and might be. Looking back, I now realise that, despite signing the serial pledge, in my heart of hearts I never truly belonged in the serial/modernist camp, any more than I had what it took to be a professional horn player. Discovering the studio was like coming home.

My increased 'leisure time' since retirement has allowed me to indulge in a few vices, one of which is listening to Radio 4, the BBC's excellent 'talk radio' channel. (Bear with me – this is relevant!) One day, I heard a discussion about education – a perennially hot political topic in the UK, where successive government ministers are forever tinkering with the curriculum and what it should or should not contain. Someone mentioned 'the three "R"s'. I'm not sure if this label exists outside the UK, but there it is always cited by those of a more traditionalist outlook as being the essential basis of 'education'. The three 'R's are, allegedly, reading, writing and (a)rithmetic. Now, apart from the obvious problem of basic literacy in respect of the letter 'R', I had often wondered why 'reading' and 'writing' were both in there, as they are strongly complementary skills, if not actually the same. In a flash of enlightenment for me, one person in the radio discussion explained that, in fact, the three 'R's (while equally compromised from the literacy point of view) actually refer to the three life skills of reading, *wroughting* and (a)rithmetic. Of course! From the dictionary, wroughting means: 'to make or do in a careful or decorative way' (as in 'wrought' iron or a carefully-'wrought' poem), and: 'to cause something to happen' (as in 'the director wrought major changes in the company').

A lot of things in my life clicked into place with this chance radio encounter, because I recognised myself. I am essentially a wroughter – a maker; a doer. My exam results at school suggested that I was fairly intelligent, but I have always felt like a bit of an interloper on this front, not least because I have always been sufficiently self-aware to know, deep down, that the nature of my intelligence does not lie in my grasp of, nor my ability to create (and then realise), grand concepts. My intelligence, such as it is, is not so much standardly 'intellectual' as *practical*, *applied*, and therefore, *pragmatic*.[2] I say 'therefore' because it seems to me that the verb 'to wrought' implies getting one's hands dirty. Vision and ideals have their place in human activity, but if you actually want to get anything done, you have to be pragmatic: you have to grab hold of the materials and shape them; interact and negotiate with them; and respond to their particular characteristics, in much the same way that sculpting implies a sensitivity to the grain of the wood or the striation in the stone.

Now, I am aware that this is in danger of becoming a confessional and I want to avoid that, but there is a key principle here that has informed everything I have ever done across my composition, performance and teaching, and that is pragmatism. I should like to explore what this actually means, and we can start to illustrate this by returning to my narrative about the studio.

4. Rethinking music: what did I learn in/from the studio?

It is tempting to claim that I am self-taught in the field of electroacoustic music – I did not take the undergraduate studio course at York, and had only the briefest of introductions to the facilities by another student. Luckily for me, Denis Smalley was by then approaching the end of his doctorate, following a year in Paris, and was prepared to spend many hours discussing *musique concrète* and the GRM with me. He even, with astonishing generosity, let me sit at the back of the studio and watch him work, which was how I acquired most of my studio technique. Nevertheless, it was the things I discovered for myself (and what does this imply about 'education' as it is so often practised today?) that had the greatest impact on me as a composer.

In basic terms, the studio turned everything I thought I knew about composing on its head. I mean this quite literally, because it made me realise – and truly (re)experience – that music is made from sound. It made me remember that 'works' are not composed from abstract structures, ideas and concepts that just happen to use sounds to articulate themselves, but that sounds take shape over time to form works. For me, at least, this means that the most successful pieces are those that demonstrate a profound link between the component sound materials and the overall form. It is the properties and qualities of the sound materials themselves that generate structure, and not the other way round. So the studio enabled me to reconnect with sound – the fundamental raw material of music – and to reconnect with it in a very direct, hands-on and sculptural way ('wroughting', again). And because I was manipulating actual sonic events, not notational representations of them, I was able to check during the process of composition that the sounding relationships were actually there, rather than simply assuming they were audible because they were visible in notation. As Trevor Wishart once said to me: 'If I can't hear it, it's not there'.

So, to summarise: I began to compose, not from the top down (as with notational approaches), but from the bottom up.

5. Reference points

There are many aspects of studio work that feed into and inform this basic approach. In my experience, the most significant considerations can be summarised under the themes of 'sound storage & access', 'primacy of the ear' and 'interactivity':

5.1 Sound storage & access

I believe the ability to record and store sound (and the ensuing possibilities of modifying it) to be the most important development in the history of music.

• It provides instant access to sound itself, not via memory or via the intermediary agency of notation;
• It allows repeated listening, which leads, incidentally, to Schaeffer's notions of the objet sonore and écoute réduite. Such privileged access is not without its dangers, however. Basing compositional decisions on fine differences that may not be apparent to the first-time listener is one such problem;
• It permits a re-engagement with fundamental aspects of sound phenomena (in my case, this led to a (re)discovery of octaves, fifths, thirds and other serial taboos).

5.2 Primacy of the ear

The ear is the means by which sound reaches the brain; composers should therefore:

• Be sensitive to the unique properties of sound materials and what they offer;
• Recognise that sound materials already imply how they want to develop/be processed;
• Be willing to structure musical time on what works in sound.

5.3 Interactivity

I should like to challenge the more usual definition of interactivity within our field, by proposing the following observations about interactive engagement in the studio:

• The constant 'testing' and 'probing' of material in a dialogue is actually interactive (in fact, I consider all focused listening to be interactive; as Nattiez points out, 'the work' is constructed not only by the composer's *poiesis* but also by the listener's *esthesis*);
• Results are assessed by, and changes made, entirely on the basis of how they convince the ear – a recursive process involving reflection / rejection / transformation / improvement / pushing the boundaries;
• 'Performance' (e.g. manipulating faders, EQ, tape recorder starts and stops, etc.) was an integral part of composing in the tape studio, even though this is more usually done today through digital surrogacy.

As an acousmatic composer, then, I work almost entirely instinctively, or by ear. Now, this makes a lot of people, especially those in academia, very jumpy. In such circles, working 'instinctively' tends to be perceived in negative terms. Because of the lack of a demonstrable 'vision', or qualifiable 'inspiration' of the composer-genius prior to the creation of 'the work' (beyond the collecting of musically promising sound materials, that is, a process which may well predate the compositional period by some time), 'instinct' is assumed to be the exact opposite of intellectual rigour. It also makes acousmatic pieces extremely difficult to analyse (instinctively composed acousmatic music is a double whammy for analysts as there is no score to allow 'out of time' access to the music). However, in my experience, working instinctively does not mean working in a vacuum, without reference to anything else; furthermore, it does not mean working without intelligence, for one's 'instinct' is clearly shaped by one's previous listening experiences – both musical and otherwise. And this listening constitutes the gathering and application of 'intelligence' in every sense of the word.

My composition practice can be characterised as a constant feedback loop in which I improvise – trying things out (timing, levels, placement, balance, signal processing, etc.) and accepting or rejecting the results on the basis of aural assessments: does this work? what would make it better? And so on. My judgements are not based on preconceived strategies, structures or formulae, and there are certainly no predetermined rules. What 'works' and what is 'right' are context dependent: they may be completely wrong in another situation. My judgements are based on close listening to my chosen material, and they are informed by all the other musical and everyday listening that I have ever done. Furthermore, my aural assessment is holistic, involving all the aspects of a sound's behaviour and energy profile at once: spectral, dynamic, spatial, etc. In my view, these characteristics are intrinsically linked, and not easily broken down into separate (and separately controllable) 'parameters'. So I am not imposing my will on sound material, but working in partnership with sound material and its unique characteristics. Together we feel our way towards the creation of a context and a final 'form'; one in which my musical – that is to say my emotional and intellectual – curiosity is somehow engaged, involved, and ultimately satisfied by what I hear.

This way of working means that I move gradually from concrete sonic events to what the piece is 'about' (the concept). Note that this is the reverse of the way that 'music' (in the western art music tradition, at any rate), with its canon of established 'geniuses', is traditionally understood to work – not least within academia. I use no (or very few) sketches or plans, and I make no pre-emptive decisions about structure (and usually not about duration unless this is imposed by a commission). All of these emerge during the compositional process, which is driven entirely by what I hear. I should add, though, that this is also a frustratingly inefficient way to compose, as I spend a great deal of time floundering about without a clue as to where I'm heading. But I see no real alternative, for to propose 'a method' would be to risk becoming formulaic. Each piece of acousmatic music needs to discover and define its own terms of reference, precisely because it is based on unique sound materials. I have written elsewhere that it seems to me that acousmatic music, almost by definition, will always be in a situation rather similar to Schoenberg's 'free atonal' period, where he really was living on his wits and literally 'making it up as he went along'. I can think of few examples from his later 12-tone period that compare with the creative energy and vitality of a work like *Erwartung*.

The underlying point of all of this is that acousmatic music – mine, at any rate – is based on the qualitative assessment of sound's unique characteristics, not the quantitative measuring of 'intervals'. And, of course, this was essentially the approach of composers of musique concrète (which, incidentally, I think is more to do with this way of working 'concretely' with sound material of whatever provenance than with any simplistic definition that implies only the use of only 'real' sounds, recorded with microphones: synthesis was an integral part of the GRM from the 70s, as Parmegiani's *De Natura Sonorum* audibly demonstrates – indeed, the interplay of recorded and synthetic sound is what that piece is about!). So you will probably not be surprised if I claim that I believe I still compose musique concrète, but now use computers and software to do it.

6. A street with two names (© B. Truax)

Much of my work weaves a drunken path down a street that, on one side, seems to be called 'Rue Pierre Schaeffer', and on the other, 'R. Murray Schafer Street'.[3] Interestingly, Schaeffer himself apparently expressed discontent with his *Etude aux Chemins de Fer* for sounding too much like railway locomotives in a shunting yard. In other words, he was concerned that the sounds were too reminiscent of their origins and insufficiently abstracted from their real-world associations. Time does not permit me to explore this in detail, but I mention it because it is important for me and my work that the acousmatic medium is pliable enough to embrace sound materials from virtually any source, and certainly from sources that lie beyond the relatively small pool previously considered 'musical'. I am talking here about the stand-off – and therefore the vast expressive potential – that exists between 'abstract', 'pure music' (whatever that is) and anecdotal reference to everyday sound materials, with audience recognition of sources as an integral dynamic of the work. I have examples of both in my own catalogue.

6.1 Works that veer towards abstraction:

Although I didn't entirely realise it at the time, my earlier acousmatic works could be considered classically 'Schaefferian'. They are not concerned with the source sounds' real-world origins, nor with their role or implications in that context, but with a musical discourse teased out of their spectromorphological (Smalley, 1997) – their abstract, 'purely musical' – characteristics. Works of mine that exemplify this approach are *Pair/Impair, Klang, …et ainsi de suite…* and *Surface Tension*.

6.2 More 'referential' works

Occasionally, however, my music would allow a glimpse of the real world to sneak in. Since the mid-90s, I have consciously exploited the original contexts (and signification) of my source sounds, to the extent that recognition of provenance has a key role in the musical structure and 'meaning' of my pieces. Even so, I was always very keen to retain a certain ambiguity of function or meaning in my music; this is certainly not phonography, soundscape composition or sound documentary. Works leaning towards this side of the street include *Sorties*, *Unsound Objects*, *Hot Air*, *Splintering*, the four works of *ReCycle*, and others.

From the late-90s onwards, my works have continued to explore this continuum between 'abstract' and 'anecdotal'. These have nearly all been multichannel – 8-channel to start with, and much larger channel counts – up to 72 in *BEASTiary*! – in more recent years. But to explain how on earth something like that came about, I need to discuss the flip-side of composition: performance.

7. The performance practice of acousmatic music

From what I have already said, you will not be surprised to hear that I regard the whole business of presenting acousmatic music in public contexts as a huge exercise in what my father used to call 'the art of the possible': doing the very best that can be achieved in the prevailing circumstances. Because of this, I tend to take the view that the tape, disc or sound file – certainly in the case of my own works – is a blueprint for potential future action, rather than a definitive statement. I fully understand and respect composers who take a different view and who maintain that what is stored on the medium is 'the work', requiring only accurate reproduction in performance. My problem is that I don't think accurate reproduction in performance is actually possible! In a straight battle between ideology and the real world, the real world will – ultimately – always win. Enter, once again, pragmatism.

What I am going to discuss here is the practice of sound diffusion – performing acousmatic music over sound systems made up of multiple (and possibly varied) loudspeakers. I am not talking about laptop performance, which I have done only twice in my life. But this experience did confirm that, for me, 'improvisation' is best done in the privacy of the studio and then subjected to the scrutiny of reflection, selection and improvement. I wish that more people would come to the same conclusion.

I said earlier that 'performance' – by which I mean shaping and moulding material in the studio; starting and stopping tape recorders at the right time; and using faders, panners and processors in a complex choreography – has always been embedded in the composition of musique concrète. Furthermore, the physical limitations of the early storage media, particularly with regard to restricted dynamic range[4] made it desirable, if not essential, to 'make the quiet bits quieter and the loud bits louder' in concert. The gestures that had shaped material in the studio were thus essentially 're-enacted' in performance to restore the profile of the work to something that carried over to a public listening context. So massaging the dynamic profile in performance is arguably as essential as manipulating 'space', which is what most people initially think of in connection with diffusion. Inevitably, if one is using multiple speakers, then their spatial configuration is a factor in what is heard. But my approach to diffusion is based on an assumption that 'space' (or what Smalley calls 'spatiality') is just one aspect of that holistic bundle of characteristics that make up a 'sonic object', and that energy in the spatial domain is likely to be strongly allied to energy profiles in dynamic and spectral domains.

8. BEAST (Birmingham ElectroAcoustic Sound Theatre)

After finishing at the University of York in 1976, and following a period of four years freelancing in London, I was appointed to a Lectureship at the University of Birmingham. This was 1980, and I immediately set about improving the Studio and building a loudspeaker system designed specifically for the public presentation of acousmatic music. After spending a couple of years getting used to this strange new world of academia, I decided it was time to get some of this music out to the public. So in December 1982 I organised a concert using the Studio's four loudspeakers together with four more of my own, plus some Motorola tweeters that I had bought. I felt that the event needed a catchy name, so I idly jotted down 'BEA' for 'Birmingham Electro-Acoustic' (I used to hyphenate the word in those days). I then thought it would be good if I could find something appropriate to complete the acronym these three letters seemed to suggest: 'BEAST'. 'Sound Theatre' seemed to fit the bill exactly. And the rest is history! (Well, no… even I would not be that pompous!) Though it is nevertheless the case that, for me at least, BEAST is, effectively, history. This is something I shall return to later, along with a few observations about being part of an academic institution.

Returning to our discussion of concert presentation, it is important to remember that the great majority of acousmatic music is in stereo. This format is, however, artificial. In our everyday lives, sound does not only propagate within a frontal 60 degree vector on the horizontal plane, but can stem from any number of positions around the listener. However, we accept stereo and feel comfortable with its limitation – largely, I suspect, because of its obvious relationship with the stage or concert platform in musical performance, and because most of the music we listen to is recorded and distributed in that format (even if we listen to it on headphones – which, technically, distorts the stereo image). Stereo is also relatively simple to understand and to set up, and is thus 'portable': everyone (in theory) knows how to play back a stereo piece. Nevertheless, stereophony is based on an illusion – albeit one that, if handled well, can be unusually convincing. Because what stereophony can do is deliver sonic images – and deliver them quite efficiently (using just two channels rather than 5, 8, or n). The sonic images I have in mind are, once again, things that I think of in qualitative terms: close and intimate; broad or narrow, dramatic and sweeping; focused or diffuse; delicate or aggressive; etc. Incidentally, sources originally recorded in stereo contain many spatial cues that can suggest particular strategies for both composition and diffusion; moreover, the fact that the encoded space is 'real' (rather than artificially created by placing mono sounds within the stereo stage) can significantly enhance the believability of images available in performance.

Starting from the standard stereo loudspeaker set-up picture in Figure 1, we know that we should be at position A if we want to hear a stereo image at its best, Whether in the studio or at home in our living rooms, we organise things so that we are in the 'sweet spot', allowing the illusion of stereo to be fully audible. These illusions permit the creation of a sound-field that exists both between and behind the loudspeakers. Sounds can believably appear at the centre even though there is no actual speaker there; sounds travelling across the image can be tracked accurately; and sounds disappearing into the distance can seem, in these relatively controlled listening environments, to move away, well beyond the walls of the actual room in which we are sitting. (Note that, in order to be believable, compositional techniques such as reducing the amplitude and the high frequency content, adding reverberation, and possibly even narrowing the image by panning it towards the centre – thereby resembling the vanishing point we all know from perspective in the visual domain – may be required.)

If, however, we now imagine that my diagram represents a performance space capable of seating 200 people, rather than an acoustically controlled studio or even a relatively damped living room (curtains, carpets, soft furnishings, bookshelves, etc), things will be very different. Even at position A (Figure 1), the dimensions of the hall, the longer reverberation time of the space and the larger distances of the listener from the loudspeakers will all contribute to a loss of detail and precision in the listener's perception of the image. And if we are not in the sweet spot, things are even worse! Off the central axis at position B, all lateral distribution and panning is distorted; too close to the

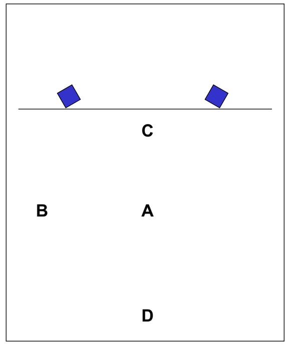

Figure 1: Stereo

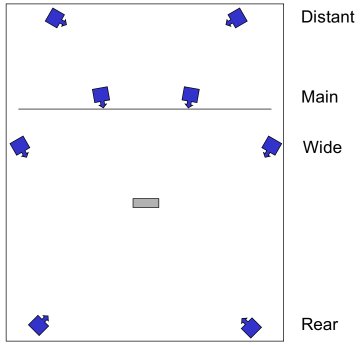

Figure 2: BEAST Main 8

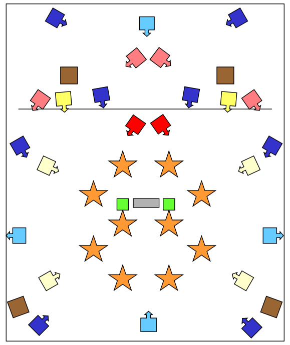

Figure 3: BEAST set-up for stereo diffusion

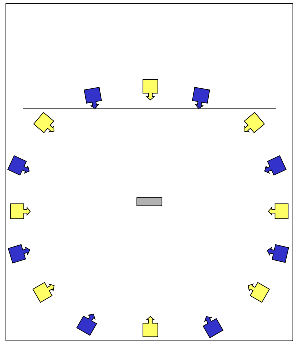

Figure 4: Two incompatible 8-channel 'standards'

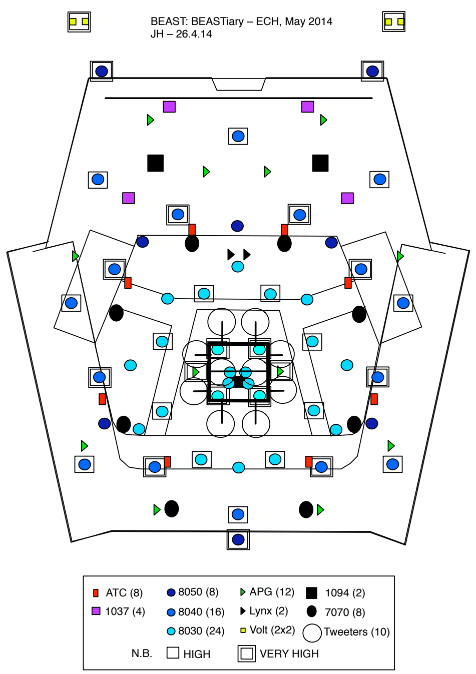

Figure 5: BEAST May 2014

in position C creates a hole in the middle; too far away, as at position D, and all the sound is distant (and probably mono!). So here we have the fundamental rationale for diffusion: in a large, public space, the original stereo images of a work are compromised unless you actively take steps to address the problems of room acoustics, audience size and position, the limited dynamic range of magnetic tape, which can seem inadequate in a public space, etc. All performance spaces are different, and there is no such thing as a neutral acoustic. (In fact, I personally find most concert halls, designed for singers and instrumentalists, too reverberant for acousmatic music.) And even if you set the levels in rehearsal, the acoustics of the space change when the audience arrives. So it is always necessary for someone to be able to intervene, to adjust what's going on, in realtime. And, of course, the ears of that person have to be in the same acoustic space as those of the listeners: the diffuser must be part of the audience.

Figure 2 shows what I call the 'BEAST Main 8', which I consider the minimum number of loudspeakers for the diffusion of stereo works. Diffusing stereo over a system like this enhances the composer's implied sonic images in several ways:

- The Distant speakers ensure that a sense of 'distance' can be accomplished, by moving the sound to loudspeakers placed in truly distant positions;
- The central location of the Main loudspeakers allow for the creation of a focused, intimate, universally audible 'central' image;
- The Wide speakers deliver dramatic lateral movement to every seat in the hall;
- And the Rear speakers allow for the effect of 'envelopment', surrounding or immersing listeners in sound.

In my style of diffusion, these pairs are not necessarily used alone. Dynamically balancing between them further extends the range of sonic images available, right up to very rapid movement across and between these eight, thereby enhancing the fast, fleeting energy characteristics in the sounds. In other words, the role of the diffuser soon moves beyond mere 'correction' to become active and interventionist. And adding even more speakers extends the range of images that can be delivered. Figure 3 shows a typical BEAST set-up of the 90s for stereo. It includes speakers on the floor, in galleries, at the mixer, and so on. BEAST in this setting becomes a subtle and extremely malleable performance tool – an instrument, if you like.

Now, one of the things that frequently happens – I would like to say 'should happen' – when composers meet an instrument like this is that it starts to influence their compositional thinking. Performance practice feeds back into composition. This certainly happened to me and it continued to happen, even as the BEAST system kept growing.

9. Public vs. private listening

Despite the fact that most people these days listen to music (all/any music) on ear-buds attached to iPods and the like, I continue to find merit in the notion of people coming together socially with the express purpose of listening to music. Despite the above-mentioned problems associated with it, there remains a continuing and thriving practice of playing acousmatic music in 'public listening situations'. However, many people (including some of my students) have criticised me for favouring the 'concert' format (i.e. people sitting in rows facing the front) over installations in galleries and other venues (though I have done those too, of course!) in which people are free to come and go and to move about.

My response to these possibly interconnected issues is twofold. Firstly, the reason I started doing diffusion seriously was to present established and new repertoire, the majority of which comprised concert works with beginnings, middles and ends. People wandering in and out at will are thus unable to hear a crucial aspect of such works: their unfolding over time. So while I have no problem in performing in galleries, I prefer to present works composed specifically for that context (it is simply a question of appropriate repertoire). Secondly, if you do not know the direction in which people's ears are facing, it becomes very difficult, if not impossible, to deliver coherent diffusion performances of pieces. This is for the simple reason that the human hearing mechanism does not work equally well in all directions: a fact that is unalterable by any fashion, trend, personal taste, or style. In very much the same spirit of 'fitness for purpose' (an example of the kind of 'admin-speak' unfortunately now so popular in universities), I feel strongly that acousmatic works fare very badly in 'club' contexts such as upstairs in a pub with people having loud conversations and ordering drinks at the bar. While I welcome attempts to broaden the audience for acousmatic music, and I genuinely believe that there is a huge potential audience 'out there', presenting acousmatic music as something it is not, and in inappropriate contexts, actually misrepresents it. Remember that acousmatic music is based precisely on the qualitative aspects of their sound materials, and therefore relies heavily on this subtlety being actually audible. Performing it in bars and clubs opens up the risk of rejection on criteria that do not even apply!

9. And then came 8-channel. . .

I was talking about BEAST – or, indeed, any speaker system – as an instrument and about how it could grow. But so far I have only really touched on stereo diffusion. With new the availability of ADAT and DA-88 machines, and sound cards with 8 outputs in the mid to late 1980s, there was a fairly serious explosion in the use of 8-channel as a compositional format. But what does '8-channel' or 'octophonic' actually mean? What is a standard 8-channel speaker layout? Figure 4 encapsulates the problem: the eight blue and eight yellow boxes represent two conflicting 8-channel 'standards' that are completely incompatible, thus presenting composers with a problem of portability: a work composed for the blue array cannot be played on the yellow array without misrepresenting its spatial contents.

I did not like either of these regular, circular set-ups back in 1999, so I stubbornly decided that I would use the BEAST Main 8 configuration (Figure 2) as my 8-channel configuration. This was largely in order to be able to place different images on different speakers: using the Mains and Wides to allow spatially detailed but wide frontal images (exploiting the area of our perception that is most sensitive to detailed 'location information') in my piece *Streams* (1999), for example; or using the Mains as a close, intimate image, surrounded by a more 'ambient' sound field in *Rock'n'Roll* (2004). This ability to deploy different materials on different parts of the array, as they are not part of a regular circle of speakers, comes from my experience with stereo diffusion, and is a major advantage of this configuration. Yet there are disadvantages, too, not least regarding what I earlier termed 'portability': the difficulty of sending the piece to other performers, promoters or festivals, since the chances are high that they will have one of the standard circular arrays seen in Figure 4. So for more practical (i.e. pragmatic) reasons – in this case, the base desire for more performances – most of my later 8-channel works have used regular arrays so as to meet the conventions of regular concert halls. Within BEAST, however, I was able to obtain the differentiation of images I was looking for by diffusing these pieces over multiple 8-channel arrays: a 'main' array, a 'diffuse' array, a 'high' array, 'close' array and so on. Luckily, a large grant enabled us to expand the system in 2004-05, and the enlarged system even enabled our pragmatism to extend to the inclusion of approaches based on idealised playback over regular arrays: these included ambisonics and VBAP domes.

Let us now turn to the issue of 'driving' a large system premised on the notion course virtually impossible to do even a simple cross-fade from one 8-channel array to another (humans do not have enough fingers), so control surfaces and software routing enter into the picture. The BEASTmulch software allows not only the grouping of multiple outputs under one fader, but also the independent mapping of inputs, outputs, faders, and much more. Indeed, almost all of the functional control aspects of a diffusion system, as well as the specification of speaker positions (for techniques like VBAP) are implemented in the system. Once again, this leads to new creative possibilities – in my own case, the idea of composing in 'spatial stems' that were intended for spatialisation in realtime during performance over a large system. This was in contrast to the act of treating a format like '8-channels' as a simple indicator of a predetermined spatial arrangement. I explored these features in my work, *BEASTory* – a 'portrait' of the BEAST system and its personnel.

But once you have, and can control, a large loudspeaker array like BEAST (now operating at 96 channels), then composing directly for the system (differentiating material types and characteristics during composition as composed stems for deployment directly onto the appropriate speakers) is an obvious next step. It is a similar approach to composing differentially for the Distant, Main, Wide and Rear speaker pairs of the BEAST Main 8 (as in my works, *Streams* and *Rock'n'Roll*), albeit on a larger scale. This possibility became clear to me during a week of testing the system in the Elgar Concert Hall, the auditorium in the new Bramall Music Building, into which the Music Department at Birmingham moved in 2012. During this week I was able to experiment with speaker locations and learn which types of material best suited which specific sub-sets of the full array. The result was *BEASTiary*. Composed for 72 channels, and performed at the opening festival of the Elgar Concert Hall (and coinciding with BEAST's 30th anniversary), *BEASTiary* is based on the same source sounds as *BEASTory*, but is developed in a completely different direction. Figure 5 shows the full 96 channels for this event, which was replicated for my final concert as Director of BEAST in 2014.

10. Back to the future… and time is running out

So here I am, looking backward, looking forward… and I am no longer Director of BEAST. This means that I no longer have 96 loudspeakers at my disposal on a regular basis, so – as a pragmatist – I'm wondering how feasible it is to continue working in the way I have described. I recently bought a new pair of high quality monitor speakers, so perhaps I shall return to my roots and start composing in stereo again. But what about teaching? Just as I am no longer Director of BEAST, I am also, apart from my last completing PhD students, no longer an academic – if I ever really was (I have often referred to myself as a 'reluctant academic', and I must confess that I took a post at a University at least in part because I could not afford to buy the equipment required to make acousmatic music in 1980). But I am approaching the end of my keynote and I have not really discussed teaching yet! On the other hand, maybe I have; teaching has been lurking underneath all of this keynote. Because I take the view, allegedly expressed by Socrates, that 'I cannot teach anybody anything; I can only make them think.'

To me, to 'teach' in the top-down traditional way would be anathema. To claim or even imply that, 'I am the fount of all knowledge and you know nothing,' is completely out of character and, anyway, is fundamentally untrue. Even though the text you are reading is based on a lecture, and therefore suggests one-way traffic in knowledge, the simple fact remains that, while you can 'teach' until you're blue in the face, 'good teachers' are only good if students want to learn.

Because of this fact, and because, as I said earlier, I am essentially a self-taught composer (and I don't have a teaching qualification either, which is now a requirement in UK universities), my approach to teaching has been simple: first, try to excite and interest students; and second, try to create a situation or context inside which they can learn. And this learning should be through experience, through experiment; through making mistakes and figuring out how to fix them. Sure, the odd bit of guidance, largely based on my own past mistakes, doesn't go amiss. But I am not trying to create compositional clones of myself. This is why my main efforts at Birmingham went into building up the studios, building up BEAST into what is nowadays known as a 'research resource', and – most importantly – building a compositional community: a partnership of equals in which creativity was enabled, had an outlet (BEAST), and in which you could pretty much guarantee finding somebody who knew about a particular piece of software that could accomplish a certain task (because I certainly didn't!). In this model, I'm not there to say 'this is right' or 'this is wrong'. If anything, I'm there to say 'I have no idea – let's try it and find out!', while also providing another pair of ears to bounce things off in an attempt to help students discover their own responses to what they hear (Socrates again: 'To find yourself, think for yourself'). I don't know if this was the right way, but it was the only way I could do it.

And it seems to have caught on (you might consider it a virus!), as I can produce a list of over 40 of my students (mostly PhD, but also Masters and undergraduate, plus occasional studio sessions with other people's students) who are currently teaching, or have recently taught, in the Higher Education sector.

Like my approach to performing and to composing, my approach to 'teaching' is pragmatic, then. I do not – cannot – set out from certainty, from a vision or from a concept, or from an all-embracing knowledge of anything. Indeed, I don't know if anyone can. I set out merely in a spirit of open-minded enquiry, to explore this astonishing universe of sound and to discover what works and what doesn't in a particular context. I am delighted to have found so many talented fellow travellers over the years, and I am sure that I have learned more from them than they have from me. So, to them, and to you for listening to me today – thank you!

Footnotes

1. I say this because it should be obvious to anyone who knows my work that I use the computer as a tool: a means to a musical end, not an end in its own right.

2. The 'applied' stands in contradistinction to 'pure' science and mathematics. I have often detected more than a whiff of prejudice against the applied forms in certain parts of academia.

3. My thanks to Barry Truax for this wonderful image.

4. About 63dB between hiss and distortion for magnetic tape, which is what I first worked with.

5. BEASTmulch was written as part of a research project led by Dr Scott Wilson, funded by the UK's Arts and Humanities Research Council: Development of an intelligent software controlled system for the diffusion of electroacoustic music on large arrays of mixed loudspeakers.

Featured Article

What is at Stake in the Politics of Digital Music Archive Access Policies? A Brief Look at Some Evolving Issues.

by Elizabeth Hoffman

If Jorge Luis Borges were brought back to life today, in the second decade of the 21st century, he would no doubt believe that his speculative proposition of a universal library – a library containing every work that had been, or could ever be written – was close to being accomplished in the form of crowd-sourced online libraries that purport to store and catalogue all of the world's knowledge. The cultural implications of these newly evolving structures are clearly vast, but this article seeks to understand them in terms of our collective concepts of Music History – particularly electroacoustic music history. How have crowd-sourced online libraries impacted on electroacoustic music curricula so far, and how in what new ways will they do so in the future? Most importantly, who gets access to these libraries, and how? The politics of access management is of vital relevance to all computer music composers working with music in recorded form, and to all composers and musicologists working with notated scores - including of experimental contemporary music. It is thus issues of access that I seek to analyze in this brief essay.

Crowd (or group) created libraries (often called "public repositories") stand in stark contrast to the many single-entity curated digital music collections that exist online, such as Ubuweb. The large number of users who upload materials do so in response to a one-time request, as per a temporally delimited archive creation process; or in response to an ongoing invite, much like Wikipedia's model. In a classic expression of the Web 2.0 paradigm, users are the content providers – albeit working in tandem with the site managers and creators. Such crowd created repositories may be conceived of as digital assemblages with historical or cultural preservation goals, or they may be community clearinghouses for sharing and exchange. Their management and access strategies therefore vary in relation to their goals.

1. Selected comparisons of curatorial strategy and maintenance

What follows is a description of three public repositories for music that each implement different access models: the Free Music Archive, the Open Music Archive, and the International Music Score Library Project.

1) *The Free Music Archive* (http://freemusicarchive.org/about) is a legal charity and it accepts monetary donations in that context. A curated site, its purpose is to foster public access to high quality digital files of legally downloadable new music of all genres, but especially "experimental" music. The FMA does not, therefore, accept all submissions. Its curators include more than a dozen international open-source sound collection and distribution entities. The FMA's principles flow from those that guide its parent, a listener-supported radio station called WFMU (Jersey City, NJ) in 2009. WFMU's Board meetings are open to the public. WFMU is run by a team of publicly acknowledged individuals. FMA's download numbers for particular postings are public.

2) *The Open Music Archive* (http://www.openmusicarchive.org/) embodies a radically different concept. It was created by UK sound artists Eileen Simpson and Ben White in 2003 as an ongoing research project to locate, digitize and distribute out-of-copyright recordings. They specialize in archiving other contemporary archives, including installations with historic materials, or sound documentary efforts. Differently to FMA and IMSLP, the site is fascinating as a musicological resource, offering critical meta-commentary about the nature of recording, preservation, and ownership. This site does not publicly reveal its user base; it promotes its files as knowledge and materiality for further artistic use, which typically manifests in compositional use by its user base. This archive seeks neither membership fees nor donations.

3) *The International Music Score Library Project* (https://imslp.org/) is a repository created in 2006 which focuses on digitized scores, although recordings and videos are meaningful components, too. In contrast to the FMA, the IMSLP is arguably more attuned to the past than the present, since it specializes in scores that are in the public domain. Yet it valuably includes 20th and 21st Century composers; Frederic Rzewski, with over a hundred self-uploaded items, is one important example. Scores for mixed music by early tape composers, including Varèse, are also present. It is thus an invaluable educational and scholarly aid – particularly for those whose school libraries may not have a physical score, or for whom properly scanning an oversized score would be a practical inconvenience.

It is worth noting that the owner of the IMSLP decided in 2015 to transform the free access forum into a two-tiered access system. IMSLP thus now invites members for $22.-/year; non-members can still access content, but the trade-off is banner advertisements and a 15-second per item download or viewing wait-time. As of this writing, IMSLP has at least one "non-affiliated" portal that does not impose the download wait for non-members – Canada (PML-CA). The IMSLP is now run by a company called Project Petrucci LLC, of Delaware, NJ, USA; and this corporation does encourage monetary donations.

2. A focused look at music access' political questions

The experience of using an online platform or service for free, only to have this use subject to restrictions or controlled via managed access, is a familiar one with contemporary digital media. Such 'bait and switch' business models succeed on the basis that users have already invested time and resources into a particular database, and so will grudgingly accept the shift in access model – but what are the politics of this shift when the content is created by users themselves? The evolution of the IMSLP's access policies prompt such consideration.

The IMSLP's maintainance itself is communally based, or 'bottom up'. Since there is no centralized curation it is arguably the most democratic repository of these three archives. Anyone may contribute virtually anything so long as the site maintainance specialists do not object on the grounds of intellectual property transgressions, and so long as the item is Music. However, the IMSLP is also the least transparent: no statistics regarding number of item-by-item downloads are available from its undeniably massive archive. In other words, balanced atop the bottom-up processes of curation is an evolving top-down political philosophy and practice. Despite being wholly dependent upon its user base, the site does not advertise its board meetings externally, nor does it reveal any other information about its long term (and recently devised) financial plan to which the IMSLP ties the membership implementation (and, implicitly perhaps, the incorporation). Looking into the future, a researcher of models for community repositories might reasonably ask the following: Does a Digital Music site initiator or manager have an obligation to the user community to ensure perpetuation of the site beyond some theoretical point of the initiator's personal interest or capacity? And is such an obligation based on: the length of time that the site has been in operation? The size of the user base? The nature of the content in relation to cultural or scientific knowledge? Finally, what are the implications of a public respository becoming privatized?

In December 2015, a comment piece by Norman Lebrecht initiated a long discussion concerning this issue. Two recurring discussion criticisms seem particularly significant to me in relation to the questions posed in this article. The first is the assertion that a co-op has been monetized after the fact, and without offering compensation to those who played a role in the database creation. The second is that the monetization makes use not only of others' manual labor, but also of their intellectual and private property – in the latter case this was done without their authorization for it to be sold.

Consider a fuller explanation of the second point. While for public domain components the contributions by volunteers are 1) their time and 2) their property, i.e., their digital files; for new music, the contributions are intellectual property that has been ceded to IMSLP "to use … in a manner similar to a work in the public domain." (This is the IMSLP's stipulation for any contributions.)

New music on IMSLP, including the category of arrangements, is often tagged with greater license specificity than is the public domain repertoire. For instance, a "Creative Commons Attribution-Non-commercial No Derivatives 3.0" provision is common. Monetizing these uploads after the fact would thus seem to disregard original wishes, the license still applying after the download but not to the download itself.

As a reader of the blogpost will note, there are also numerous comments that are not critical at all. Two focus points are a defense of the "reasonable" nature of a mere 15-second wait time, and an interpretation of the monetization itself as a creative idea. "Let [the initiator] reap the benefit of it." In sum, a more fine grained analysis of feedback would be required to analyze the demographics of negative, neutral, and positive responders regarding IMSLP's new access policies.

3. Conclusion

This article has sought to ponder the politics and philosophy of preservation goals and access in public repositories, beyond the explicit or implicit social and economic choices that regulate them. These choices contribute to the shaping of our contemporary digital life – they impact us as a professional community, as individual composers, and as non-specialist users. Through the mediation of public repositores, new notions of authorship, ownership, authenticity, access, canonisation, and value systems are being imagined and implemented. Can privatized sites retain their commitment to the ideologies of openness and knowledge sharing that characterised the sites when they were public, or are they being fundamentally transformed?

Digital music archives now reach millions of diverse users across the globe. How users respond to particular digital archive models will have profound impacts on how the archives persist and evolve. This in turn will impact on how we teach music courses, how we program concerts, and how we define and tell our histories (as well as Herstories!) Music repositories can contribute to the redefinition of expertise, as less advantaged individuals are granted access to resources that were previously reserved for the wealthy or those with institutional affiliations. Do private access models undo some of the good work achieved by peer-produced models like Wikipedia, or do they improve these services?

Communal archives offer remarkable opportunities for musicians. They have the potential both to educate and to encourage independent thought, cultivating users as cultural participants, social activists, and consumers. What better way to reactivate concert audiences than to encourage online outreach, participation, and cultural engagement?

Footnotes

1. A discussion thread at this link appeared last year: http://imslpforums.org/viewtopic.php?f=1&t=8187 [site accessed 2/6/16 - 6/7/16].

2. "Musicians are made to wait as free score site goes pay-for", attracted a large number of comments early on (the article and its discussion thread may be read in full, here: http://slippedisc.com/?s=musicians+are+made+to+wait&submit=search

The First NYC Electroacoustic Improvisation Summit, New York City College of Technology

Thursday 27th Febrary, 2016

by Eric Lyon

For a while now, mainstream computer music conferences such as the ICMC have faced a curatorial challenge, as computer music has become increasingly varied in its scope and has achieved near ubiquity in its means of production. It has become difficult to highlight a particular research agenda or compositional direction at these events because the conference is quickly swamped by the sheer variety of research directions in play. While the resulting smorgasbord of ideas and music, along with an essential community-building aspect, ensures the importance of the ICMC and similar conferences for the foreseeable future, it is now largely the role of smaller events to bring focus to thematic directions of particular interest.

This curatorial impetus has been met admirably well by a new event called the New York City Electroacoustic Improvisation Summit (EIS), conceived of and directed by Kevin Patton and Adam James Wilson. The inaugural EIS took place at New York City College of Technology on February 27th, 2016. The focus of this summit was instrumental improvisation in interaction with computer systems that themselves provided improvised structures and signal processing.

The role of improvisation in computer music has an interesting history. We define computer-based improvisation as music in which the computer improvises or responds to the improvisation of a performer in real-time. Different inputs lead to different outputs, which is sharply distinguished from the "instrument and tape" model in which the output from the computer is fixed and irrespective of the musical behavior of the live performer. The slow CPU speeds available when Max Mathews wrote the first acoustic compilers at Bell Labs during 1957-1966 precluded computer-based improvisation. Instead, a compositional framework for computer music was established in which music is programmed and compiled to a fixed medium outside of real-time. As microprocessors and personal computers became available in the 1970s, ensembles such as the League of Automatic Composers began to create live, improvised, networked computer music performances. The publication of the MIDI 1.0 standard in 1983 greatly accelerated exploration of live computer music, which often had a large improvisational element. Notably, most of this work was centered around the affordances of the MIDI protocol, which allows for organizing musical structure at the note, harmony, melody, rhythm, and instrument level, but affords little control over sample-level DSP.

At the same time, there was an intense focus on developing the possibilities of DSP in mainstream computer music during the 1970s and 1980s, resulting in important breakthroughs such as Frequency Modulation, LPC, and FFT-based processing. So there was a kind of bifurcation for a time in computer music between non-real-time, composed, DSP-focused music, and real-time, improvised, musical pattern-based music. However even as early as 1980, one can see a trend toward increasing interest in live, microprocessor-based music, when reviewing the titles of the papers from the 1980 ICMC.[1]

In the decade of the 1980s, arguably the most ambitious computer music improvisation project was George Lewis's *Voyager* (1986-1988), which features a computer-based, improvising expert system that analyzes and responds to live improvised input from human performers (or even from itself). As the 1990s progressed, a couple of important transitions occurred. First, increasingly fast CPU speeds enabled a transition from MIDI (and the relatively unadventurous sounds provided by commercial digital synthesizers), to live digital synthesis, where the accumulated power of computer music research into audio DSP could be increasingly leveraged into live computer music performance, which often had a significant improvisational element. At the same time, as I've argued elsewhere,[2] computer musical timbre research seems to have hit a plateau in the 1990s, creating space for a redirection of computer music research efforts that, I believe, still remains to be fully acknowledged and acted upon. One such space is computer-based improvisation, which brings us back to the EIS.

While electroacoustic improvisation is not necessarily limited to computer music, at the 2016 inaugural edition of EIS, a decision was made to program exclusively computer-based improvisation. This curatorial decision led to a focused program of improvisational computer music works that demonstrated a broad range of musical expressivity, while validating the proposition that computer-based improvisation is a musical category worthy of attention.

Chapman Welch's *500 Great Things about Wichita*, performed by Brandon Bell, commenced with vigorous on-body percussion strikes on chest and legs, which was quickly joined by a delicate, computer-generated harmonic accompaniment. The work then transitioned to a series of short sections, each characterized by performance on a single percussion instrument with autonomous computer-generated accompaniment, based on live sampling of the percussionist. The eloquent and structurally convincing decisions made by Bell, combined with the freedom afforded by the improvisational context, made this a lovely and satisfying musical offering.

Clip Mouth Unit, a duo project of Dafna Naphtali and Jen Baker performed with a high-energy mix of Baker's trombone interjections and Naphtali's intense yet urbane vocal stylings, combined with varied and unpredictable computer-generated textures and live processing of the acoustic sound, all presented with a comic's madcap sense of timing. Despite a wide range of surprising musical swerves, the performance never lost focus.

My *Parallel Noise Construction* was composed for the new music violin duo String Noise. One of the violinists, Conrad Harris, was out of town, so I performed his part, with Pauline Kim Harris on the other part. The work is a noise-driven improvisation in which a program generates dual sets of improvisational performance instructions, while also randomly assembling different signal processing algorithms through which the violins are processed. During the sound check, Kevin Patton performed my part on violin so that I could listen from the main hall. His improvisation was intense, and also quite different than mine, or Kim-Harris's. This suggests the intriguing possibility that at another electroacoustic improvisation summit, performers need not play their own pieces, but rather could swap into performing through someone else's system.

A Bird Escaped From the Snare of its Fowler by Kevin Patton and Nikki D'Agostino combined D'Agostino's hyper-intense saxophone playing with a more deliberate music coaxed from the computer by Patton, based on real-time analysis of the saxophone performance. D'Agostino's improvisation had some fine lyrical moments that nicely balanced the initial mode of intensity that dominated the performance.

Eighteen Eighteen performed by Adam James Wilson and Arto Artinian unleashed frenetic, heavy rock stylings performed by Wilson on electric guitar, and an intense keyboard backdrop performed on Haken Continuum by Artinian, all mediated by an oracular listening and improvising program written by Wilson. At times during the performance when a spooky third voice hovered, I was reminded of the mysterious third that walks always beside you, as described in T.S. Eliot's The Waste Land.

Tattoo of a Gesture by Margaret Schedel stood out at in its use of a printed score that integrated both textual instructions and precisely notated rhythms. Christopher Howard contained the manic expanse of composer-provided possibilities within a taut, obsessively controlled, and increasingly virtuosic performance. While computer processing was clearly audible, particularly in live filtering of drum sounds, the main sonic focus was on the percussive sounds produced by Howard.

Solo for Voice and Computer composed and performed by Paul Botelho reminded of what an incredibly intimate instrument the human voice can be. In this delicate improvisational duet, Botelho seamlessly merged his live voice, an exquisite countertenor, with a live-generated texture built from sampling of the voice, and initiated by interactions with his laptop computer keyboard. Botelho cannily integrated expressive physical gestures into his performance, particular of the hands and arms, making his occasional human-computer interactions seem completely natural. The expressivity of the performance seemed both the point, and completely impossible to notate.

Through the aesthetic success of the first EIS, Patton and Wilson have provisionally validated their proposition. They now face a wealth of possibilities to explore in the next EIS. Will the range of electroacoustic improvisations be broadened beyond computer interaction to embrace analog electronic systems? Will invited musicians workshop their systems for the public? Will members of the public have an opportunity to experiment with featured improvisation systems? Will the performances be broadcast to the Internet, or archived online? Will telematic improvisation be incorporated? Will improvisation systems with no humans in the loop be presented? Patton and Wilson have already made a serious contribution to computer music with their first EIS. It will be quite interesting to see what direction they choose with the next one.

Notes

1. http://quod.lib.umich.edu/i/icmc/bbp2372.1980?rgn=full+text

2. https://www.researchgate.net/publication/298981852_The_Future_of_Spatial_Computer_Music

ICMC 2016 Concert Reviews
Utrecht, The Netherlands

Thursday 15th September
Robert Henke - Lumière II.2
TivoliVredenburg Grote Zaal
21:30 - 22:30

by Lauren Hayes

Lumière II.2 is an evolving piece, composed in hardware and software, heard in sound, seen as light, and played out over dozens of performances in sites that range from castle courtyards to industrial spaces. The work is synæsthetic – at least in metaphor – by not quite producing involuntary experiences in a secondary modality, but offering beautifully coupled audio-visual phenomena. At ICMC 2016 this took place inside the Grote Zaal, being the only performance of the conference held in TivoliVredenburg's grandest space.

The theme of ICMC 2016 was 'Is the Sky the Limit?' and Robert Henke introduced *Lumière II.2* with a discussion of the role of limitations within his own creative practice. In this case, these were the limitations of the technology; the limitations of what can be achieved with a given number of lasers; the limitations of the control systems which forced Henke to create his own software in order to achieve his artistic goals; the limitations of our perceptive capacities and sensory systems; and perhaps the limitations of working with a technology often awkwardly associated with trance clubs and laser harps.

We were guided through an audio-visual Euclidean topology of points and grains, lines and waves, planes, Bowditch curves, and symbolic signifiers, which further developed out of the bounds of the screen into three dimensional constructs and columns (made visible by the use of smoke). The monochrome palette grew into an array of colours, culminating in a striking red circle which was stamped emphatically on the screen with a suitably cinematic accompanying sonic gesture. The suggested interpretations of this moment from the audiences members I spoke to afterwards were both visceral and colourful.

With limitations come boundaries and edges, at and around which perhaps the most interesting situations can occur. Aside from the impressiveness of the rigorously constructed audio-visual material that was presented to us over the course of the piece, I was drawn to the spill of the laser projections onto some of the stage lighting above the screen, where the quietly dormant objects of the theatre became unintentionally animated. Similarly, there were a few moments where the trajectory of a moving line appeared to jump off the bounded canvas onto the nearby wall, allowing me to speculate on the agency of the instrument itself.

Henke's music is described as 'on the edge of contemporary club culture' [1], yet when the techno-flavours appeared in the music, I wondered about another limitation: the limitation of the concert hall which forces its audience to forego the shared participatory experiences of moving bodies. I think back to the inspired choice to feature a standing-only performance from Luke Abbott at the Sonorities Festival of Contemporary Music at SARC, Queens University Belfast in 2013. When Henke offered the audience an improvised encore, he finally gave us permission to move around and also take recordings on mobile devices. Of course, allowing the latter during *Lumière*-proper would have disrupted the efficacy of the visual presentation, but witnessing audience members change vantage points, crowd around Henke's table to peer at his screen, and quietly yet excitedly converse with one another suggested that we can continue to push the boundaries of how we present computer music "without the stultifying trappings of concert society" [2].

Notes

1. http://roberthenke.com/interviews/bio.html

2. See: Garton B. 1994. Why I Hate Concerts. ARRAY: the Quarterly Publication of the In-ternational Computer Music Association, Summer 1994. http://sites.music.columbia.edu/brad/writing/papes/Why_I_Hate_Concerts.html

Pandora Concert 2
Tuesday, 13th September 2016
TivoliVrendenburg Pandora
19:00-20:30.

by Jonathan Higgins

Pierre Alexandre Tremblay's asinglewordisnotenough1 opened this packed evening concert with a bang. A cacophony of lilting rhythms bounced around the speakers to great effect before subsiding into a softer synthetic drone. As the drone progressed, bass stabs reminiscent of the opening rhythms began to develop alluding to a return of this material. Although this return was anticipated, when it happened Tremblay still managed to catch me off guard and the overall effect was incredibly satisfying.

The next piece, *dototo.006* by Masatsune Yoshio, was fantastically spatialised, enveloping the audience and filling the concert hall. Despite the density of the sound materials, particular sounds clearly occupied their own spaces within the room. Although at times the heavy use of granulation did lend itself to technological listening. Overall, the gradual fluctuations within these granular textures were excellently crafted and fascinating to listen to.

Taking the audience on a journey, Yu-Chung Tseng exploited the plasticity of recorded sound in *Between Points*. Expertly blending a series of eclectic sound materials together, Tseng's work felt reminiscent of montage soundscaping. Each material merged seamlessly into the next, creating an ever evolving sound world. Between Points was a fantastic piece both musically and technologically.

Przypadek by Michael Lukaszuk placed every day sounds like crisp packets and fizzy drinks being opened within an abstract computer generated sound world. Ambient metallic melodic fragments washed across the concert hall, gradually building in rhythmic density to create undulating textures. The piece was very well diffused with sounds seemingly moving upwards as they progressed giving the piece a terrific sense of height.

The penultimate piece was *Drops and Ripples in Spacetime* by George Nikolopoulos. Inspired by gravitational waves, the sound materials interacted transforming each other to create sonic ripples. Starting off with relatively sparse sounds the piece built in density over time as more materials were transformed. The composer was unable to attend and as such the piece was not diffused. This was unfortunate as the piece would have benefitted from being able to ripple across the space.

The keynote speaker Åke Parmerud closed the concert with *La vie Mécanique*. Despite having been written in 2004, the piece felt just as fresh and exciting as the other music on the programme. The piece focused on driving rhythms which built in complexity over time. At times the rhythm would drop away before coming back full force, a technique similar to those used in electronic dance music. Parmerud defused the piece magnificently and he was nearly as exciting to watch perform as the piece was to listen to.

Tuesday, 13th September 2016
Off-ICMC
TivoliVredenburg Cloud Nine
23:00-00:00

Tarik Barri opened the concert with *Versum*; a synaesthetic audiovisual journey through a virtual universe of his own design. Creating and exploring planets and stars within the universe on the fly, the performance was improvisatory in nature. However, despite this the performance purveyed a clear sense of form with sonic materials developing, evolving and interacting throughout the duration of the piece. A wash of hypnotic FM bell arpeggios and wonky evolving beats worked in tandem with the psychedelic visuals to create a relaxing yet brilliantly engaging performance.

In stark contrast to the relaxing *Versum*, Thomas Ankersmit's *Homage to Dick Raajmakers* was a brutal barrage of harsh noise. Screaming high pitched drones punctuated with deep analogue thumps left half the audience running for cover within the opening minute. The audience that remained were treated to a highly disorientating, exhilaratingly masochistic experience. Occasionally the bombardment would subside into brief moments of respite. These were in many ways the tensest parts of the performance, leaving you wondering with a mix of excitement and dread about what would hit next. The piece ended with Ankersmit leaning over and switching off his equipment mid drone, the ensuing silence pressed on the ears before the audience erupted into a well deserved round of applause. Thomas Ankersmit's performance was captivating and his control of the Serge modular synthesiser was nothing short of masterful. *The Homage to Dick Raajmakers* was personally my favourite performance of the week.

Book Review

Peter Elsea
The Art and Technique of Electroacoustic Music
A-R Editions Inc. 2013

by Laurie Radford

A-R Editions' Computer Music and Digital Audio Series has provided many titles over the past decades that focus on computer music analysis, composition and research. Many of these titles continue to serve as important study guides and reference. Peter Elsea's 2013 contribution to the series, *The Art and Technique of Electroacoustic Music*, adds a title that attempts a broad overview of the concepts and technologies employed in electroacoustic music production and performance. Elsea is well known in the Max world for his widely read MaxMSP and Jitter tutorials and for his LObjects collection of Max objects.[1] He is also known to more than three decades' worth of students as Director of the Electronic Music Studios at the University of California, Santa Cruz from 1980 to 2013. A visit to his (retirement!) webpage provides a glimpse into his teaching activities, research and compositional output, and a sense of his personal engagement with the computer music community. His tutorial page begins with this statement: 'These are tutorials I have written over the years for various courses in Max. These papers are usually written in a hurry, so errors inevitably creep in. [Heck, errors are inevitable in papers written slowly, and things like books which get reviewed dozens of times before publication.]' One is advised to keep this statement in mind when reading through the 500+ page The Art and Technique of Electroacoustic Music as some errors have indeed "crept in."

In the Preface, Elsea chastises other books about electronic music and audio production for their exclusive coverage of science and technology. Yet, *The Art and Technique of Electroacoustic Music* focuses almost exclusively on the 'technique' in the book's title with little discussion of 'art', little mention of the composers and practitioners that work with the techniques covered, or representative repertoire and performances that they produce. The book resembles a lab manual and covers an enormous breadth of material. Perhaps the 'art' referred to here is that of the apprenticeship and mastery of a technological skill set as an art unto itself? The author provides some justification for this by stating that most of the repertoire of electroacoustic music is available online or via some available media. Unfortunately, the single reference

media. Unfortunately, the single reference
to these resources is to emf.org which no longer handles audio sales. A better choice might have been electrocd.com which provides an extensive offering of historical, recent and new electroacoustic releases. The author identifies a number of target readers for the book and by doing so provides a good indication of the objectives, contents and the strategy of presentation: composers wishing to move beyond pre-packaged sounds and production environments, who strive to expand their skills to incorporate advanced and powerful recording and sound transformation/generation techniques, and who wish to develop their technical and musical listening skills. Therefore, it would have been beneficial to provide some guidance to the practitioners and the work accomplished in electroacoustic music (in an appendix or as part of the Resources for Further Study), especially given the author's identification of a potential self-learner readership for the text.

In addition to providing a compendium of diverse information and guidance for conceptual and technical issues in electroacoustic music, the book also serves as an illustration of Elsea's pedagogical method(s) in the area with his 'custom textbooks for each course' clearly serving as the foundation of many chapters. It also, for the most part, represents the academic and home studio experience of the 1990s and 2000s. As such, new practices such as DIY, circuit breaking, and telematics performance that have emerged during the publication of the text are not considered.

The book offers 19 chapters that cover six areas, including: Building the Studio, Fundamental Concepts and Techniques, Music Store Electroacoustic Music, Synthesis, Research-Style Synthesis, and Live Electroacoustic Music. Building the Studio opens the text with a general overview of the main considerations in creating a suitable environment for electroacoustic music production. One finds this type of introduction at the beginning of many books on audio production and sound recording and it is an important area to explore in such texts, especially those that are targeted at the self-learner. In this case through, there are terms such as 'band-limited' and 'scrub' that are introduced in this first chapter that are not explained in sufficient depth until later chapters, a fact which reduces the effectiveness of the immediate discussion and may even cause some confusion for those employing the text as an introduction to the discipline. Another example is this statement in another early chapter: 'many of the features in professional-grade mastering applications are not essential to composition'. Mastering is never again mentioned in the

the book. The assumption that a neophyte reader will understand what is meant by 'composition' (later referred to as 'pure composition') is also problematic given that the concepts of sequencer/loop style mixing techniques on one hand and sound exploration, montage and transformation on the other are mentioned in passing early on in the text without clear historical or social definitions.

This calls into question the intended genre of music making under discussion and the target audience for the book. On one hand, the compositional exercises and suggestions seem to assume an exploratory avenue of creative work as a norm. One the hand, the prevalent use of conventional pop music terms and concepts such 'bass line', 'backing track', 'lead line', beginning-middle-end, and a prevalent concern for tuning and pitch a well as conventional rhythm in sequencing, belies a kinship with popular music writing and production. (The statement 'in electroacoustic pieces the concern is usually more about getting pitches to match in the first place' is puzzling but also revealing in this regard.) The mix of basic sound recording and pop music production terminology with conceptual and composition advice that arises from more exploratory strains of electroacoustic music results in a lack of clarity in regards to the type of music and sound and music production the book is in fact discussing. Then again, it could be read as an attempt to cover the range and breadth of practices gathered under the electroacoustic banner, to erase the lines between genres which in fact currently employ many of the same software and hardware tools.

Part 2 consists of a series of chapters that serve as introductory guides to working with sound. These cover the basics of acoustics, sound recording, sound processing, audio mixing and useful references to general compositional applications of some of these concepts and techniques. Components of sound, recording technologies and equipment, audio editing techniques, a host of processing types including EQ, compression, reverberation, distortion, modulation, the digital audio workstation and audio mixing are discussed with clear descriptions and ample illustrations for conceptual reinforcement. By necessity, the discussion of such a great many concepts is introductory; yet they are clear and orderly in presentation. As with a number of areas in the text, some terminology, for example 'dither', is employed in passing and not sufficiently defined. This could prove problematic for those coming to the discipline for the first time and the inclusion of a glossary could have solved this issue and provided a useful reference component for the title. (That said, a quick online search for most of these ill-defined terms will serve the same purpose now.) The tutorial origins of some of the texts include step-by-step instructions for software interface use. At a time when young and new users are reasonably adept at the use of computer interfaces in general, the plodding nature of some of these instructions (i.e. a lengthy, step-by-step guide of how to use a transport control on an audio recording application) seems unnecessary.

Part 3 and 4 of the book cover a variety of software and synthesis concepts and make a distinction between 'Music Store' and 'Research-Style' modes and cultures of technology-based music making. The Music Store section consists of seven chapters covering MIDI, Sequencing Programs, Samplers and a series of chapters on various synthesis methods. It is somewhat puzzling that the discussion of FM, additive, spectral, granular and modeling synthesis are housed under this Music Store rubric given that all of them originated in, or at least have been highly developed at research centres and were subsequently taken up by commercial enterprise for wider exposure and distribution. The discussion of MIDI and sequencing in this part of the book is extensive and covers details of the MIDI protocol, typical MIDI studio routings, the main parameters and interface affordances of a MIDI sequencer as well suggestions for efficient and creative use of a MIDI-enabled environment. The discussion of various aspects of MIDI messages and routing could have benefited from the typical schematics for clarity, and the discussion of hardware versus software synthesizers takes a disconcerting turn when plugin synths are introduced without any explanation of what they are or how they function.

A concise chapter on Samplers offers an introduction to the historical origins of sampling instruments and many of the conventional parameters and functions that have been implemented in these instruments over the years. Two topics that receive extensive discussion and tutorial treatment in this part of the book are Voicing Synthesizers and FM Synthesis. The author uses Absynth as an example of a typical softsynth and discusses its architecture, menus, oscillators, filters, and envelope generators in detail but in a general enough fashion to be applicable to most other synthesizer modules. (As mentioned later in this review, the discussion of software and hardware synthesizers, divided into separate non-adjoining chapters in the book, seems unnecessary and potentially confusing to first-time users.) FM Synthesis receives the most detailed discussion of any individual topic in the book. FM is obviously an important technique in the author's palette given the comprehensive and clear presentationof of FM generation, control and use. Yet, the statement that FM synthesis and sampling make up 90% of the synthesized sound in use today may be true for the latter, but perhaps not so much for the former. Analysis/resynthesis, additive and granular synthesis, phase vocoding and image mapping techniques for synthesis, as well as physical modeling are only briefly discussed at this point relegating these powerful and by now quite well-known techniques to a novel category that interested readers will hopefully further explore in other appropriate literature.

One wonders if parts of *The Art and Technique of Electroacoustic Music* will be of use in the not too far off future given the many technical issues stated as fact that, in fact, are no longer such: the use of Protools (an old version at that) as the paradigmatic example of a DAW given the current wide-spread use of many other, more diverse products by younger electronic creators and performers; the statement early on in the text that there are MIDI sequencers and then there are audio DAWS and that someday they might be combined when, in fact, this has been the case for most products for decades; the restriction of a discussion of automation to gain changes in a mix without mention of the extensive affordances provided by automating many, if not all, parameters of plugin processing; the suggestion that 'the practical low end for widely distributed music is 60Hz' at a time when current practice in numerous genres of electronic music worships prominent compositional components below that frequency and the market is flooded with affordable subwoofers; an almost flippant approach to spatial design as an integral component of electroacoustic music ('pans are set at the start and seldom moved again'; [the reverb] control…will probably not move during the mix'); and the inclusion of products such as the 'Walkman' as contemporaneous with the iPod! One suspects that Elsea was well be aware of the period-specific nature of many of these statements given a comment featured on his website, 'I use published texts from time to time, but they become dated quickly, and of course can't address the unique aspects of these studios', and chose to present his tutorial materials 'as is', a testament to his long, dedicated career to teaching electroacoustic music. From one perspective then, The Art and Technique of Electroacoustic Music acts like an auto-ethnography, foregrounding the personality of the author amidst the concepts and techniques that he obviously cherishes.

One of the most successful sections of the book is the introduction to 'Research-style Synthesis' methods including Common Lisp Music, Cmusic, Csound, the Composer's Desktop Project, Rtcmix, Supercollider, Impromptu, Pd, Csound, ChucK, and MaxMSPJitter. Most of these are only mentioned in passing, often in reference to their connection to the historical lineage of Max Mathews' Music series that kick-started music computing in the 1950s. The last three programs cited receive a more thorough overview and an introductory tutorial. The basic functions and syntax of Csound and ChucK are discussed and accompanied by ample code examples that provide a point of entry for those interested in investigating these powerful open source programs. The author also succinctly covers some basic computing concepts (variables, operators, library functions, loops, arrays, unit generators, Markov chains, etc.) that provide a framework for the computing skills required to employ these programs. The real-time potential of live-coding in Csound and ChucK is mentioned, but especially in the case of ChucK, greater emphasis and illustration of its live-coding affordances should have been offered. In addition to the static examples provided, a short example of a live work flow in ChucK could have been included as an example of this growing area of live electronic music practice. The many online video examples of live-coding in action could have been mentioned since they provide a much clearer illustration of this practice than anything a mere description can do.

Elsea is well known for his MaxMSPJitter tutorials that many new (and experienced!) users of the software have visited online for guidance. The chapter on Programming with Boxes and Lines draws upon these succinct tutorials and provides a basic introduction to the program and its basic functions covering most concepts and details that a fledgling Max user would require to get started. The substantial changes to the program since Version 7 are not reflected in the discussion or illustrations given that the reference here is Version 5. Nonetheless, the fundamentals of control flow, routing and timing in Max, principles of audio, recording and synthesis in MSP, and 'A Hint of Jitter' and interconnections between image and sound provide a rapid flyover of this paradigmatic composition and performance environment.

Upon initial reading, it seems somewhat puzzling that a chapter on programming Synthesis in Hardware would follow the lengthy introduction of code-based methods, especially given Elsea's opening statement: 'Is hardware dead?' As noted above, a discussion of both softsynths and hardware synths could have been combined and provided a clearer picture of the commonalities and differences between them. And yet, this chapter may very well provide valuable advice for exploring the hundreds of abandoned synthesizer and sampler modules that are on offer at low prices at many pawn shops and online second-hand sale services, and that feeds the cyclical, retro phenomenon of current hybrid electronic music practices. The lengthy discussion of the Kyma system in this chapter, enhanced by several illustrative video documents, serves as a representative example of a self-contained hardware-based composition system; but the claim that it is used by most professional electroacoustic composers is exaggerated and misleading. The space would have been better dedicated to a discussion of the many currently available control surfaces and tablet-based systems that are fusing the paradigms of software and hardware in the studio and on the stage.

The last two chapters of the book discuss details and issues regarding Live Electroacoustic Performance and Composing for Electronic Performance. The first of these begins with a short paragraph implying that the acousmatic tradition is a thing of the past and was 'never popular', and that audiences for electroacoustic music remain few and lack patience for anything that doesn't provide conventional markers of liveness in performance. Given the dizzying number of electronic and media festivals, acousmatic concerts, emerging live electronic music practices, journals and texts discussing these creative activities, and the substantial support for these events by audiences all of the world, it is difficult to take this view seriously, and undermines much of the subsequent practical advice that is offered. The chapter proceeds with a brief survey of conventional sources of control for live electronic performance (keyboard, guitar, wind, string, percussion) and devotes a mere third of a page to circuit-bending and one page to the world of NIME. A discussion of the author's teaching activities employing piezo transducers offers a brief glimpse into the engaging and undoubtedly inspiring pedagogical atmosphere he maintained throughout his teaching career. It makes one wish that more examples like this, drawn from more than three decades of experience, where included in the text. The final chapter provides a cursory survey of several main paradigms of Composing for Electronic Performance including classic instrument plus tape, instrument plus processing, as well as some of the practical and notational issues involved. Some examples of capturing performance data such as pitch and tempo via MIDI for player control of processing and temporal aspects of a performance are discussed and illustrated. Pitch detection and score-following functions for live performance are mentioned (with promising recent research in these areas not considered), and several examples of employing random number generation and distribution procedures in Max for use in

distribution procedures in Max for use in
performance (once again via MIDI) brings the chapter and the book to a close.

The words of wisdom for which Elsea is known online (and in the classroom) lose some of their coherence when combined in this lengthy and sometimes rambling tome. Qualitative observations and critique about products, practices and concepts remain very personal throughout the text, and draw upon the author's lengthy and intimate experience with the creative objectives and technologies under discussion. That personal touch lends an inviting tone to the text, as if by reading it one can still take a class with Peter Elsea! The check list of practical tips for "Putting Your Show on Stage" that concludes the chapter is the advice of an experienced practitioner.

The wealth of audio examples provided on the accompanying DVD contributes to the usefulness of the text for teaching and learning. A variety of single sounds, synthesis examples, signal processing examples (including video examples of actions in plugins and other software interfaces), brief sound etudes, and comparisons of various audio characteristics and situations are provided to illustrate examples and discussions throughout the text. At this point in time, the sound and video examples would be more useful if they were available in an online repository, especially given that many users no longer have access to CD/DVD drives. Most chapters conclude with suggested Exercises for exploring the concepts and techniques introduced and Resources for Further Study are provided as a starting point for further reading.

Returning to Elsea's webpage statement regarding 'errors creeping in' and 'books which get reviewed dozens of times before publication', it would have been beneficial if A-R Editions had followed this advice to guide some of the content and discussion to a more succinct, focused and up-to-date state. Nonetheless, with some judicious and up-to-date guidance, suitable supplementary materials and information, both technical and historical, as well as links to practitioners and repertoire, T*he Art and Technique of Electroacoustic Music* could serve as a principal or reference text for an introductory undergraduate electroacoustic music composition and techniques course or for the self-learner who wants a bird's eye view of the electronic music terrain.

Notes

1. See http://peterelsea.com/maxtutorials.html)

2. http://artsites.ucsc.edu/ems/music/PQE/More_PQE.html