NOISE FROM THE UNDERGROUND

Selected Texts 2009--2024

Daniel McKemie

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℗ VCR Publications

Foreword by Ryan Ross Smith

*For Willie, Roscoe, Johanna, Mort, Pauline,
and Dean, and all of my musical collaborators.*

Foreword:

Forward

– Ryan Ross Smith

Preface

In 2017, I was set to return to music school for doctoral study at the University of California, Santa Cruz. Instead, I chose to retract my fellowship and stay in New York, a decision I have never once regretted. Still, I wanted to be challenged musically and academically, so I made a promise to myself to continue the research I had proposed and to eventually collect it all together in the future.

This book was born the same way so many of my other projects have been, as a DIY adventure rooted in fun and personal fulfillment. I'm not aiming to make any grand statements or peddle a manifesto. It's a story, a path explained, ideas shared, and half a lifetime of experience gathered.

Everything in this book is a nod to the feeling that keeps things moving forward, whatever it is that drives us to create, to paint something yet to be seen, to record something yet to be heard, to share thoughts about what we experienced that day.

As I write this preface in 2021, it is around the time I would have been finishing my doctoral thesis had I chosen to pursue it. This book stands as the completed promise I made to myself four years ago. It isn't as polished as a doctoral thesis, nor is it meant to be; instead, it stands as a document of selected essays, program notes, and text pieces, presented as-is.

Many musical ideas from this period are intentionally not included, as they are still in progress. Notable examples include work with the New York Deaf Theater, various dance troupes, the *Pneuma* series, and numerous tours, records, and shows with bands all around the country. Thank you for reading![1]

Daniel McKemie
December 2021
Brooklyn, NY

[1] Find supplemental materials and ongoing updates at my website: `danielmckemie.com`

The Las Vegas Story

During my last year in Las Vegas, in 2009, I met a lifelong friend and artistic collaborator named Theodore Dourbet (Teejay). We briefly shared a house, and it was there that our ideas began to soar. Our garage was outfitted with an assortment of instruments and tools, including a 1990s-era direct-to-CD recorder that we used to capture our work.

We spent a lot of time talking about music, and there was one question we kept coming back to: "How do we make a record where the listener can begin with any track, listen on shuffle, and still have a consistent experience?" All these musings culminated in a collection of records, concerts, and scores. The first of our projects was an album called *Negativland*, released under the name *Kloned Noise for the Underground.* The name is a combination of our respective monikers: *For the Klones* and *Noise from the Underground.* This stands as one of the most interesting and uniquely weird album releases I have been a part of. The instrumental parts were recorded solely on Casio keyboards. The audio clips were taken from an episode of “Saved by the Bell” in which the gang is in the mall trying to acquire tickets to a U2 concert, as well as other choice quotes from the show’s cast. We edited all of this into a collage in a style inspired by the band *Negativland*, who were sued by U2 in the 1990s for unlicensed sampling. We layered this collage over the Casio jam, split it into ten tracks, and found that they could be played in any order and still achieve the same musical narrative.

Around this same time, Teejay showed me a drawing of intersecting lines that he had seen in a dream and wanted to realize as music. Using my familiarity with alternative notation systems and graphic scores from my time in music school, we got to work on exploring what we could do. We assigned one sound to each line. We decided that the x-axis would represent time, then chose a variety of assignments for the y-axis and interspersed them throughout the page. The realization became *delsdic 33109* and was used in a film piece we worked on together later that year.

The experience of working with Teejay’s drawing inspired us to put together a set of music by John Cage. I had just finished my recitals, papers, and lectures for my degree, so I was looking for my next project. We picked a few Cage works that could be performed without the need to read a score, to suit how Teejay had a solid understanding of avant-garde performance practices but did not read notated music. The program included *Imaginary Landscape No. 5*, *4'33"*, *Radio Music*, and a portion of *27' 10.554" for a Percussionist*. We constructed recordings and rehearsed regularly, with Teejay adding a great theatrical effect to the music through the use of costumes and tiny percussion instruments.

We had no plans to perform the set live, but we were approached by a local promoter to play a First Friday at a downtown Vegas bar called The Bunkhouse. First Friday is a monthly event in downtown Las Vegas where local art galleries open their doors to the public to exhibit works. It also includes ad-hoc street performances, graffiti stalls, and eclectic vendors. The event quickly grew into an established affair, with satellite parties at bars throughout the area. The promoter exclaimed with great enthusiasm that he was a fan of my music and loved John Cage. I tried my best to explain that the set was quite challenging for audiences and might not fit within a larger bill like First Friday. He kept insisting and asked us to open the night.

With about 150 people in attendance, we were the prelude to an evening of DJs, slam poets, and hip-hop acts. A perfect fit! We printed programs and placed them throughout the bar, hoping to encourage the public to inform themselves and form their own opinions about the music. The crowd's reception was predictable. Most were slightly intrigued, disgusted, bored, with a few engaged. Not so different from a concert hall.

The conclusion of our set was our realization of *Imaginary Landscape No. 5*, played over the PA via compact disc. With three minutes left in the performance, the promoter, who once again so dearly loved John Cage,informed us that he was pulling the plug. Teejay and I continued sitting and acknowledging the performance in progress. After the show, the promoter berated us: “I didn’t think you would just press play on a CD player and sit there,” and “None of that even made sense.” To this day, I’m honestly unsure if he truly knew Cage’s music, if he just wanted us to play, or why he thought it made sense to book us for a First Friday hip-hop night. No matter the reason, it remains one of the most memorable shows I’ve ever played. We later performed the same set at a local café, where it was very well received.

Sonal Mirror Whitewash

The piece is to be performed in a crowded area where talking and conversing is encouraged during performances, such as a bar, concert hall lobby, or the outdoors. The idea is to distort and filter the sounds that already exist in the venue and feed them back into the audience. The audience is entirely responsible for the source material used in the piece, the performers on stage are entirely responsible to carry out its form.

Place any number of microphones around the room. Good places to put them would be near tables of conversing crowds (asking their permission ahead of time would be a wise idea), near trash cans, behind the bar, etc. However, NO microphones should be placed on the stage or used by any of the performers for any reason. If a microphone must be used, they must leave the stage to do so.

Run all audio sources into effects processors. Any effects in any amount may be used. Employ high, low, and band pass filters, delay, reverb, EQ, panning, etc.

The form of the piece is carried out by the performers. One example could be to start minimal by utilizing one effect, then increase, move onto another effect, go back to the original effect, and so on. Create a gradual build that will climax at the conclusion of the piece, where it should end abruptly.

Absolutely NO instruments are to be used by the performers. But if an audience member has an instrument, then they may play it during the performance, so long as it takes place offstage.

As an option to further determine form, the onstage performers may project pre-recorded audio into the audience. They should not process these projections in real time.

Daniel Steffey
March 2009
Las Vegas, NV

Near the end of my last year in Las Vegas, I wrote *Sonal Mirror Whitewash* (2009), the first piece of mine intended for others to play. It is a text score that aims to break down the performer/audience barrier, not a new idea but new to me at the time. I wrote it after being asked to play a somewhat left-of-center bill at the Freakin' Frog. For the on-stage portion, Teejay and I were joined by our close friend RJ McBain. For the audience participation part, we set up microphones around the bar at tables, trash cans, in the middle of the room. Wherever they could reach. I handed out copies of the score as a kind of program, an idea lifted from our Cage performances, and coupled it with a brief explanation of what we were about to do. It was the perfect number of people for a first run of a piece like this. We performed for about twelve minutes until the promoter walked on stage and said to me, as politely as possible, "Hey man, this is too weird for this crowd, and they aren't really into it. We're gonna have to pull the plug on this one." Meanwhile, I was observing a few dozen people paying close attention to what we were doing and actively taking part in the shared performance space.

I was starting to get used to this routine, but this time I wasn't so understanding of his decision to end my set. The sound cut off abruptly, and I yelled for everyone to applaud the promoter as he shut it down. I was over it. I was over my hometown and trying to do anything new there.

We packed up our gear and began to leave, but I saw a small group of people who were genuinely excited about what they had just witnessed. This excitement eclipsed the fact that we had been rudely interrupted. It was satisfying to feel that I had done something artistically important to me, and I was genuinely happy that it meant something to others, even if those who booked the gig disapproved.

Daniel McKemie
January 2021
Brooklyn, NY

Radio Pieces (2009)

This is taken from my first master's thesis, extended program notes/short analysis of a suite of short works for radio. I share it here as an illustration of my practice and aesthetic that would be present for the decade following. They can be heard on my record Chloros, released on Edgetone Records in January 2011.

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The *Radio Pieces* were conceived over a one-month period in August 2009, when I first arrived in Oakland with nothing but time to kill waiting for classes to begin at Mills College. I found myself with limited means of making music which were just some random bits of hardware, a computer with no music software, and a radio. Most of my music up until that point had centered on sample manipulation, but none of the equipment for this was available to me. This situation sparked an approach that has stuck with me ever since: making music with the means and equipment available at a given point in time.

Many people believe that making quality music requires the best gear. But great equipment doesn't make creative music—creative people do. In this case, I downloaded Audacity, the open-source DAW, plugged in a radio, recorded the sounds between stations, and began to work with them. The radio I used was a Model 1750, 17-transistor, multi-band radio built in 1965, with AM, FM, Shortwave, and Very High Frequency (VHF) bands. It was the first time I used a radio as a compositional tool in my own music, despite earlier interest and exposure to it as a musical instrument.

My interest in radio stems back to early childhood, as I was always drawn to its 'non-musical' sounds like static, frequency sweeps, beeps, whirs, hums, and other noise. During my undergraduate studies, I became interested in the music and philosophies of John Cage. Through performing his percussion works, I naturally developed an interest in his other compositions, such as *Credo in US* (1942) and *Radio Music* (1956), both of which use the radio as a musical instrument and compositional device.

In *Credo in US*, written for percussion quartet, Cage uses the radio as both a sound source and an element of indeterminacy. While it would be another nine years before he began using chance operations to compose, indeterminacy first appeared in this work. In the score, Cage only indicates when the radio should be on and its dynamic level—there is no specification of frequency or band. At the time, the only widely accessible bands to the public were AM and shortwave, as FM had been invented in the 1930s but was not made commercially available until after World War II. I performed the radio/phonograph part in this piece on my final undergraduate percussion recital. While exploring the range of sounds available on the AM band, I was once again struck by the sounds of radio that had been ever present in my childhood.

Cage's *Radio Music* functions differently. This work is scored for one to eight radios, with one player per radio, and is divided into four sections that can be played with or without silences between them. The score is written on loose-leaf sheets of paper, with numbers indicating frequencies in the AM band and lines indicating silence. The events were composed using chance operations, and the use of radio introduces an indeterminate quality so that no two performances are ever alike. While my own work with radio does not align with Cage's specific approach to indeterminacy and chance, it does share the same underlying unpredictability, which is using radio as an instrument places the sound material completely outside the composer's control. That lack of control gave me great comfort in working with such a chaotic medium.

While Cage's music helped me rediscover and appreciate radio, my approach was more deeply influenced by the music of Karlheinz Stockhausen, namely his works *Hymnen* (1966–67), *Spiral* (1968), and *Kurzwellen* (1968). Each of these pieces could warrant a thesis of its own, but I will briefly outline how they influenced my work.

Hymnen is a work conceived in four movements that Stockhausen refers to as "regions," constructed from recordings of national anthems and shortwave radio broadcasts from around the world. The sonic palette is incredibly complex and fascinating. After meditating heavily on this piece for the entire summer before moving to Oakland, I realized there is an entire sound world at our fingertips, one that exists within the radio itself. Stockhausen's masterful work was never something I sought to imitate, but it helped me understand that there is a wealth of sound material all around us, and that I no longer needed to wait and listen for sound to come to me. The sounds are already there, right in front of us.

The two other works, *Spiral* and *Kurzwellen*, were composed immediately after *Hymnen*. *Spiral* is written for any solo instrument and shortwave radio receiver, while *Kurzwellen* is scored for piano with shortwave receiver, electronium with shortwave receiver, large microphoned tam-tam with shortwave receiver, one viola with a shortwave receiver, and a sound projectionist with two filters and four potentiometers. This instrumentation reflects the ensemble Stockhausen was performing with at the time the piece was written.

These two pieces call for the players to improvise around the sounds of the shortwave radio according to a specific set of instructions and a graphically interpreted score. The draw for me lies in the core of the work: the act of performative phrasing on instruments against and alongside the sounds of the radio—and, in the case of *Kurzwellen*, with each other. This music laid the groundwork for my first compositions utilizing radio and continues to be a major influence on my work today.

Radio Piece No. 1 began as an experiment in editing sounds and arranging them in ways I found pleasing. No extra processing was done, and after being satisfied with the result, I decided to continue and create a series of studies involving different manipulations of radio waves. Six more pieces emerged from these experiments, each titled sequentially in the order they were created. For every piece, new recordings of radio sound material were made.

Radio Piece No. 2 adds manipulation through editing. This movement explores changes in playback speed—slowing down and speeding up the audio. Most of the sonic material comes from the VHF range of radio waves, which includes local weather reports, police and rescue channels, and air traffic control broadcasts. The latter can be heard clearly at 1'39" into the work.

Radio Piece No. 3 introduces more effects like phasing, tremolo, pitch shifting, and delay. Initially, I felt this piece was clumsy, cheesy, and simply not very good, due to its inherent lack of control. But in context with the rest of the movements, it fits well. This piece was crucial in shifting my thinking toward a cohesive work with a clear direction, rather than just a collection of studies.

Radio Piece No. 4 is centered on intuition. Realizing that these experiments were evolving into a larger set, this movement was designed as a breather. It employs time stretching and downward pitch shifting of the samples and was composed during a period when I was listening to a lot of Mahler symphonies.

Radio Piece No. 5 is the longest piece in the series and the only one that features live manipulation of sound, as opposed to mapped-out, fixed manipulations. Initially, I felt the piece was a great success but didn't align with the spirit of the others, since it was performed live. I later found this concern to be misguided and chose to follow my instinct. This became a major turning point in my compositional life, as I began to see that laying down a fixed plan before composing did not always serve the music. While I'm a fan of algorithmic music, I don't enjoy composing it. The most important outcome of *Radio Piece No. 5* was that it sparked my interest in spontaneous and performative approaches to electronic music. This is further explored below in the later work *(R*Sin)F=M*.

Radio Piece No. 6 is where I imposed strict limitations, focusing entirely on a single point on the dial. All audio was taken from the 148 MHz frequency in West Oakland, California. The goal was to showcase the synthetic-sounding, chaotic nature of the radio on its own. What might seem distorted, manipulated, or artificially created is, in fact, a naturally occurring phenomenon—one that many people overlook every day.

Radio Piece No. 7 features recorded audio sped up multiple times, resulting in a short, glitchy, and mostly quiet piece with qualities distinct from the rest of the series. A kind of coda. There is an accompanying video work titled *Film for Radio* that visually reflects its aesthetic[2].

Daniel Steffey
May 2011
Oakland, CA

[2] This video was subsequently lost after a hard drive crash and YouTube flagging copyright infringement because of the music used. I lost the appeal, even after proving ownership of the music, and the video remains stuck in cloud purgatory.

Instructions on How to Write Music

...or DIY Pieces Nos.1-11

These pieces serve as instructions on how to compose, or realize, a piece of music. The media in which they can be realized is always open to those realizing it. Any instructions or musical indications not given, are up for interpretation to those realizing the pieces. For example, if no tempo is indicated, the tempo is free; if no pitch is given, the pitch is free; and so on. . .

These are NOT performance scores alone. These pieces are to be written down, or electronically realized, in advance. As they stand, they do not serve as guidelines for performance, unless you are quite adept at composing on the fly. Sometimes parts may be vague or unclear, in these cases it is encouraged that you interpret these instances the way you best see fit, or that suits you most.

"Ideas cannot be owned. They belong to whoever understands them."

- Sol Lewitt

Daniel Steffey
June 2016 - April 2017
Brooklyn, NY

DIY Piece 1

for Ryan Ross Smith

Four voices

One cycle = 11 beats, repeated 11 times (121 beats)
Piece duration = 11 cycles (1,331 beats)
Tempo = 140-240+ BPM

12 universal timbres are chosen in advance, with each assigned a different number:

- *Voices 1-3:* short sounds, 1 beat in length, each a different timbre.

- *Voice 4:* long sound, 3-9 beats in length, a single timbre that is different than any timbre in Voices 1-3

Beats can either be assigned a timbre or a silence, but every sound must occur on a beat. Timbres may repeat within a cycle. An example of a few cycle realizations are as follows:

Cycle 1

- *V1* = A different timbre per beat chosen by the realizer and the timbres are repeated at will. Silence is assigned to every other beat.

- *V2* = First 5 beats (1-5), on timbres 2, 3, 5, 7, 11 (chosen at random, do not repeat within cycle, no silences) Last 5 beats (7-11), on timbres 1, 8, 4, 10, 9 (chosen at random, do not repeat within cycle, no silences) Middle beat (6), on timbre 6

- *V3* = One random timbre chosen between 1-11 assigned to each beat, with no silences

- *V4* = long sound (again, different than any timbre in Voices 1-3), 3-9 beats in length (chosen at random), occurring every 18-38 beats (chosen at random)

Cycle 2

- *V1* = The same timbre on every beat, with no silences

- *V2* = A different timbre between 1-11, occurring only on beat 1 of each repetition

- *V3* = Complete silence

- *V4* = The same timbre occurring for 9 beats, with 18 beats of silence in between each event

Cycle 3

- *V1* = The same timbre every third beat, with silences in the remaining beats

And so on. . .

Once 11 of these cycles are completed, the piece is over.

DIY Piece 2

for Ralph Lewis

Any number of voices

This piece is in five parts, for any number of voices. There are indications of *low, medium,* and *high*. These are assigned to whatever parameter the realizer chooses. The duration of the piece is 13 minutes. The duration of each of the five parts is as follows:

- Part One: 1’36"
- Part Two: 3’25"
- Part Three: 2’19"
- Part Four: 4’34"
- Part Five: 1’06"

Each part has this many notes or sounding musical events (i.e. placement of dynamics do not count as an event) taking place:

- Part One: 27
- Part Two: 62
- Part Three: 74
- Part Four: 189
- Part Five: 32

These events should be split into low, medium, and high. This distribution is chosen at random. You must have at least ONE event in each category. For example, in Part One, you could have this distribution:

High: 3
Medium: 13
Low: 11

OR THIS:
High: 18
Medium: 5
Low: 4

The placement of each event is determined by the realizer, so long as the events follow these instructions on where they belong in the space. Again, the events may be distributed among any number of voices.

DIY Piece 3

for Nick Wang

For four voices (each voice has four 'sub-voices')

The structure is in four parts, totaling 8 minutes.

- Part One: 0'28"
- Part Two: 1'40"
- Part Three: 3'35"
- Part Four: 2'17"

A voice is an entire system of contained parameters unified by a single timbre. A sub-voice is assigned a musical parameter that must hold through each part and may be reassigned in different parts. An event, in this case, is any change in activity at all (i.e. change in: rhythm, pitch, dynamic, technique, knob turn, etc.). The voices themselves should be composed independently, each having their own system of pitch, rhythm, volume, or whatever other parameters the realizer chooses to implement. For example, in V1 a change in the sub-voice controlling volume in V1 counts as an event in V1 alone. As in you cannot have a change in volume in V4 and mark it as an event in V1.

- V1A = pitch changes
- V1B = dynamic changes
- V1C = rhythm changes
- V1D = technique changes

Conversely, sub-voices may be contained systems on their own. In this case, here is an example you may have:

- V1A - all pitch, dynamics, rhythm, technique; in one system
- V1B - all dynamic, range, filtering, technique; in one system
- V1C - sim., as above
- V1D - sim., as above

Again, these assignments must remain consistent throughout each part, and can change when a new part begins.

The number of events in each sub-voice, for each part, are as follows:

Part One:

- V1A - 35
- V1B - 17
- V1C - 10
- V1D - 7

- V2A - 36
- V2B - 38
- V2C - 24
- V2D - 49

- V3A - 46
- V3B - 15
- V3C - 25
- V3D - 25

- V4A - 12
- V4B - 19
- V4C - 19
- V4D - 34

Part Two:

- V1A - 58
- V1B - 18
- V1C - 2
- V1D - 61

- V2A - 6
- V2B - 71
- V2C - 78
- V2D - 49

- V3A - 50
- V3B - 69
- V3C - 69
- V3D - 94

- V4A - 34
- V4B - 36
- V4C - 32
- V4D - 45

Part Three:

- V1A - 95
- V1B - 27
- V1C - 43
- V1D - 62

- V2A - 26
- V2B - 83
- V2C - 43
- V2D - 82

- V3A - 67
- V3B - 95
- V3C - 25
- V3D - 20

- V4A - 64
- V4B - 80
- V4C - 13
- V4D - 68

Part Four:

- V1A - 99
- V1B - 2
- V1C - 79
- V1D - 39

- V2A - 100
- V2B - 46
- V2C - 49
- V2D - 42

- V3A - 48
- V3B - 71
- V3C - 6
- V3D - 11

- V4A - 22
- V4B - 19
- V4C - 1
- V4D - 101

CA State Route 24

During the summer months of 2009 when I first moved to Oakland, I spent a bit of time each day scouring the classifieds for gigs to make extra money. I came across a listing for a composer and sound designer for a theatre piece to be performed at the Berkeley City Club, a stunning hotel on Durant designed by Julia Morgan. The listing was pretty bizarre and immediately caught my attention. After a few phone calls and a meeting, I landed the audition to write and perform the live score.

The Stone Wife is a play written and directed by Helen Pau. She runs a company called the "Nec Nec Lab", though I can't say whether the company ever produced anything else or if it's still active. The production was a surrealist adventure involving clowns, actors, a puppeteer, Butoh, composed lighting, and prepared piano.

Helen had originally envisioned the score for timpani, marimba, vibraphone, piano, trumpet, and snare drum. But due to space constraints, budget limitations, and the fact that I didn't play trumpet, we settled instead on snare drum, bowed saw, auxiliary percussion, prepared piano, and electronics. Bit by bit, the score and cues came together, and she gave me a good amount of creative freedom.

One scene called for performing Franz Schubert's *Moment Musicaux No. 3* while a puppeteer, standing on a ladder, mimed a pair of tiny shoes dancing under a spotlight. This was preceded by a clown, center stage, launching into a monologue about needing a heroin fix.

This was my first time writing music for theatre or doing any kind of sound design work. I drew obvious inspiration from John Cage when deciding on prepared piano sounds and shaping the placement of silence. I learned quickly that when writing for theatre, the composer has to be both sonically and physically aware of space: how sound travels, how it sits in a room, and how it interacts with movement, lighting, and speech. I gave myself two simple rules:

1. Don't play too much.

2. Don't play too loud.

Being mindful of the physical space is a bit more abstract. It means putting yourself in different spots around the theatre and thinking about your intentions in each one. Where do your cues land? Where do they go? Where is everyone else in relation to that? Not just in terms of time, but in terms of space.

The baggage I brought to writing music for *The Stone Wife* wasn't all that different from what was happening in my music on the other side of the hills. *Annabel Lee* was my first composition where I aimed for complete control, with every step carefully planned. Not with a serialist approach, but as a heavily structured effort. There were only written notes. To its detriment, I took a very classical approach to the piece, which ended up teaching me an incredibly valuable lesson.

It was written for a workshop recording session at Mills College for cello, bassoon, baritone saxophone, and voice, and centered on exploring counterpoint in low-register instruments.

Roscoe Mitchell was my primary composition teacher at the time, and after the recording session we discussed the piece. I told him I thought of it as a complete failure. I said I absolutely hated the music, and that I never wanted to subject anyone to having to listen to or play something like that ever again. It was a tremendous disappointment.

I had brought this piece to my lessons over the course ten weeks, and I asked him why he hadn't been more critical of it. His response: "Because you were really into writing it, and you had a vision. I didn't want to disrupt that vision." He went on to explain that it's important to realize your ideas and follow through with what you have in mind, but just as important to recognize when to let it go. He said the most important thing is that you're doing the work, not just talking about it.

Roscoe has a common saying, "Don't let the grass grow under your feet."

That was the point when my relationship to formalism and tradition started to change. It shifted from a sense of practice, to instead a respect for tradition.

By the end of my time as a student at Mills, I felt like I might have just started to figure something out. Written for the Mills Contemporary Performance Ensemble, conducted by Steed Cowart, *Concerto for Contrabass Clarinet and Nine Musicians* explores the radical timbres of each instrument, especially the soloist. It wasn't about controlling the compositional landscape, but instead getting to the sounds I found most interesting.

The solo part includes loosely notated, blaring multiphonics, a quasi-soloistic amplified violin, chaotic piano clusters, and loud percussion with a strong sense of emergency conveyed through tam-tam and hand siren. I set out to write a piece with no clear sense of musical release, and if release did exist, it was up to the listener to find it.

This piece grew out of a growing dissatisfaction with much of the music I was hearing at the time. The conversations around aesthetics often turned into strategies for how best to appeal to audiences. The concerto was a direct reaction to that. Why take a generally inaccessible style of music and force accessibility?

Not to say, “Let’s damage the audience and make them unhappy,” but rather, let’s stop being so concerned with the opinions of those who go to concerts. If someone doesn’t like the music, they can take a note from the Europeans: leave the concert, boo incessantly, or just do something else.

I’m always grateful and humbled by those who come to listen, but it’s more important to be honest than to be well liked.

Daniel Steffey
August 2013
Los Angeles, CA

DIY Piece 4

for William Winant

Two voices
Tempo = 60 bpm

The sound events of each voice should be short, percussive, and clearly defined. The piece exists in three parts. Each voice functions in the same way but separately. The length of the three parts are proportionate to one another like this:

- Part One: 1/4 of total length and is a continuous group of notes without rest
- Part Two: 1/2 of total length and is two repetitions of Part One, with rests replacing some notes
- Part Three: 1/4 of total length and is the first repetition of Part Two with more added rests

Each voice is realized independently but with this same method. Part One is the basis for the other two parts. Once all of the events are introduced, the remaining two parts repeat those series of events, with silences replacing the sounds.

To start, the realizer must pick a length of time for the ENTIRE piece. This should be measured in the number of beats (as opposed to minutes and seconds) because the numbers of events determined in the following method is done per beat. Through random processes, the realizer determines the number of events in each beat. The events in each beat are equally spaced at all times. For example, if a beat has three events, the events are played as a standard triplet would be played. The number of events that are possible in one beat range from 1 to 9.

For Part Two, the same sequence of events from Part One is repeated two times with rests replacing some attacks. To apply rests, the realizer uses random processes to first choose how many rests will substitute the events for each beat. For example, if a beat has 5 events, the realizer chooses, at random, numbers 1-5. Say, 3 is chosen. This indicates that 3 rests will substitute events 1-5.

Secondly, the realizer then chooses at random WHICH events will be substituted. So again, the realizer chooses at random, numbers 1-5. Say, 1, 3, and 5 are chosen. In this instance, in a beat with 5 events, beats 1, 3, and 5 are substituted for silence, leaving only beats 2 and 4 respectively.

This version of Part One with some attacks replaced with rests is played twice in a row Part Two.

x = original sequence of events
y = original sequence WITH substituted silences

Part One = x
Part Two = y + y (the same result, simply played twice)

For Part Three, the result of one iteration (y) of Part Two is taken and more silences are substituted for events. A similar process is executed as it was to realize Part Two from Part One. The number of silences to be substituted per beat is selected at random, but the range of silences is concurrent to the number of events in the beats, as they exist in the iteration of Part Two. To clarify:

- In Part One, one beat has 5 events.
- In Part Two, said beat is substituted with 3 rests.
- In Part Three, the beat from Part Two is taken, but the range is 1-3.

The difference in the process for realizing Part Three, is that the specific events that are substituted with silences are up to the realizer, as opposed to being randomized.

Note, that the language of rests and silences is used interchangeably. The requirement is that the rests/silences MUST BE EXACTLY THE LENGTH of the event in which they are substituting.

DIY Piece 5

for Dino Ayala

One voice

This piece is in one part that alternates between two groups of sounds: A & B:

A = a continuous, unbroken sound
B = VERY short, percussive sounds (irregular pulses)

The number of alternations (one alternation being A + B) is left to the realizer (or the number chosen through random processes), each alternation is different in nature.
The length of each part of the alternation is chosen at random, in terms of seconds. This ranges from 15" to 60". To clarify:

- A's length is randomly set to be 15" to 60" in length

- B’s length is randomly set to be 15" to 60" in length

On the next alternation, this process is repeated, giving new lengths to each part.

The sound quality of each part is directly relative, within that part. Meaning:

- A can change characteristics (timbral, pitch, etc.), so long as it is continuous, with no silence.
- B is to share ALL or MOST timbral qualities with A

The timbral qualities can change with each alternation but they MUST stay consistent within each alternation.

A = continuous sound of any character/timbre created by the realizer
B = Pulses. The number of pulses for that specific alternation is chosen at random each time. The range of possible pulse events during the entire part is 10-50. The pulses are irregular in nature, placed at will (or at random) by the realizer, however they should not be heavily front or back end loaded in the timeframe assigned to that part.

DIY Piece 6

for Ernesto Carcamo-Cavazos

Two voices

Tempo = 60 bpm

The ‘bursting’ voice enters at regular intervals of 12 beats between attacks. The time between each ‘bursting’ voice attack is labeled as one sequence. The ‘bursting’ voice is a sharp attack, with a fast decay, a little sustain, and a long release (although some time can be taken from the release and put into the sustain, so long as the sustain is not longer than the release). The total duration of the release should last for at least 7 beats. Because this will be very quiet, this length is flexible. The ‘bursting voice’ should be a harmonically rich sound with many conflicting elements within its spectrum, perhaps to the point of being noisy.

The pitched voices respond to the burst with a sharp attack, long decay, random sustain (see below for the construction of pitch content), and shorter (but not sudden) release that fades to nothing. The length of this sustain is chosen at random between 5-9 beats. Loudness and volume among the voices is relative and should be matched accordingly, so as to achieve evenness in all voices and one does not overpower any other(s). A piano matches this envelope, so it is an acceptable use for this kind of voice. Unpitched percussion is not. Use your best judgment to determine if the instrument can faithfully perform the assignments given by this music.

The parts are distinguished by the change of characteristics of the many pitched voices. The ‘bursting’ voice stays consistent throughout all of the parts. All parts are of equal length, comprised of full (12 beat) sequences, and are at least 72 beats each in length. The length of each part should increase based on how many more pitched voices are used, but this is ultimately left to the realizer.

Part One: All notes attack together.

Part Two: Attacks are no longer on the beat and become staggered. All of the attacks occur within the first beat (beat 1) of the sequence. The placement of each pitched voice's attack within a beat is chosen at random. The most effective way to determine placement, is to evenly divide the beat (into 4, 6, 8, etc.), and choose the number division in which the attack is to take place. For example, in more conventional music terminology, if the beat is divided into 4, the realizer chooses at random, a number 1-4, 1 being 'on' the beat, 2, being the 'ee,' 3 being the 'and', and 4 being the 'uh.'

Part Three: Pitched voice attacks become more staggered, and occur within the first 3 beats. The process for realizing the placement of attacks is the same as Part Two, only the timeframe is expanded. The length of the entire event for each pitched voice is now 2-9 beats, but the ADSR envelope still holds proportional.

Part Four: Within this parts' sequences, all voices (other than the unchanging 'bursting' voice) alternate between sounding and silence. The realizer chooses voices are sounding or silent in each sequence at random using this order of actions:

1. Whether the voice will be sounding or silent.

2. The length of the voice for that sequence.

3. Where the attack will take place: it can be anywhere in the sequence as long as the length of the event does not overlap into the next sequence. (To clarify, if the voice is determined to sound for 5 beats, then the attack cannot begin at beat 9, your range of where the attack is, lays between 1-7 (7+5 = 12).)

<u>Part Five:</u> The same series of events as in Part Four, but the events can now overlap sequence boundaries. To clarify, as stated in Step 3 of Part Four, 'given that the length of the event does not overlap into the next sequence,' no longer applies in Part Five.

<u>The Construction of Pitch Content:</u>

The construction of pitch content is done by the following steps. The realizer must create a fixed system of proportions, unique to the media of the voice. The range and its divisions (registers) are chosen at the start and cannot change at any point throughout the piece.

First, the realizer must determine the divisions of each pitched voice's range. To begin:

1. A series divisions of range is proposed. For conventional instruments, the range can be split at the octave. To use the piano as an example, the lowest octave can be considered register 1, the highest octave is register 7 (or 8). To use an electronically generated sine tone as an example, your range can also be divided at the 'octave,' but must be proportionate. To clarify, 0-110Hz is the lowest register, 14,080-28,160Hz is the highest register. These range can be divided in any way so long as they registers of equal size according to how you are dividing it. It is also worth stating, that the low/high boundaries of the instrument(s), or electronics, do not have to be the extremes. Regardless, each register within the voice is assigned a number, in the order of lowest to highest.

2. If the realizer wishes for the voice to play in a limited range throughout the piece, they may choose to do so. The range boundaries (the overall low and high

pitch available on the instrument) may also change throughout the piece, but the boundaries stated in one part must hold throughout that part, and may change with the next part. Under no circumstances, can the register proportions/assignments themselves change. To clarify, if the octave is your assignment of range for the piano, you cannot change it to the fifth for the next part.

3. The register in which the voice plays, for each sequence, is decided at random.

Next, the pitch content within that register must be decided. This is done using a similar process of determining register. To begin:

1. The register is divided equally to determine pitch which values of pitch to exist. To use the piano as an example, the octave is the decided range and it is split into 12 parts, giving us the chromatic scale. Each pitch is assigned a number, from low to high (in this case 0-11).

2. The pitch that will sound during that sequence is chosen at random. Once the specific pitch or frequency has been determined, it CANNOT change within that

sequence. For voices on instruments capable of playing multiple pitches simultaneously, i.e. the piano, this process must be repeated for each pitch to take place with the sequence. To clarify, one key that is played on the piano is treated as its own voice. Every additional key thereafter is treated as its own voice. A separate register and pitch determined for each. The realizer must take precautions when deciding the voice placements of a single instrument so as to not write impossible passages.

The steps of the process should look like this, using the piano as an example. Pitched Voice 1 Realization of Sequences in Part One:

1. Register assignment is the octave. The division of the octave is 12.

2. The length of the event to take place, randomly chosen between 5-9 beats; realizer picks 7.

3. Register boundary is the middle (C3-C5)

4. Randomly choose which register plays (3, 4, or 5); realizer picks 5.

5. Within register 5, randomly choose the pitch to sound (0-11); realizer picks 11; B-natural is the sounding pitch.

The event for Voice 1 in the first sequence of Part One is that the piano player strikes and holds B-natural 5 for 7 beats, before releasing the key.

The process is then repeated and changed accordingly based on the description of each part.

Reminder About ‘Bursting’ Voice:

The ‘bursting’ voice contrasts with all of the other voices. It is a consistent sound that is unchanging throughout the piece. For best results, it should be a harmonically rich sound in which the spectrum is clearly heard (i.e. an oboe ‘burst,’ would likely not yield interesting results). Special care should be taken for an unpitched ‘bursting’ voice; while some unpitched materials could have interesting effects, they should generally be avoided. This also holds true for ‘noisy’ timbres. But this is not an absolute. The requirement of the other voices on the other hand, is that they must be capable of sounding pitch.

A Combined Practice of Two Seemingly Disparate Musical Archetypes

In music academia, improvisation and classical performance have traditionally been treated as two separate practices. The study of improvisation is often relegated to jazz and world music courses, while notated music remains firmly embedded in the classical domain.

My first instrumental work, *Chloros II* for Amplified Violin and Percussion (2009), was born out of a series of improvisation sessions between myself and violinist Christina Stanley. We would improvise to explore our respective instruments together, searching for new timbres, forms, listening practices, and musical ideas. *Chloros II* also fulfilled the requirements of an assignment for a composition seminar I took during my first semester at Mills College. Each student was tasked with writing a five-minute composition for a concert, and the purpose of this piece was to codify those improvisation sessions.

This was the only assignment for the course, aside from showing up. The course itself was exactly what a seminar should be: meeting in a room, presenting work, sharing ideas, engaging in open discussion, and providing and receiving honest critique. The form of the piece was very rigid, with the violin part moving between notated and improvised passages, while the percussion part remained fully notated throughout to help define the musical time.

When I presented the idea for the piece to the class, it was met with unexpected bewilderment from both the teacher and the students. The primary concern was how I would find a violinist capable of both reading music and improvising. I explained that Christina was fully capable of both, but the response from the group was that the piece would struggle to get future performances because of these seemingly disparate and unrelated musical skills.

The only response I could think of was, "Well, it's just not my problem if you can't play the piece. If the piece is too hard, practice to learn it, or don't play it." A pragmatic approach I found effective during my years as a performer. Like so many pieces of new music, it has so far only seen one performance.

The blending of traditional notation, graphics, improvisation, and written instructional elements in my music is something that has kept the process fresh for me as a composer. But still, why was the idea of a classical player who could also improvise considered so outlandish? I share this experience for those who may be interested in combining written notation and improvisation, whether as composers, performers, or both, to encourage the practice of both paths and to intertwine them at the points they find most meaningful.

Having played in many bands over the years, I've heard plenty of arguments downplaying the value of notational literacy—none of which were particularly strong. Some feel they don't want to be tied to the notes on the page, as if knowing how to read music obligates you to always use it. Others worry it will inhibit their voice as a performer, as if understanding the alphabet somehow limits one's ability to speak.

My music theory professor at UNLV, Dr. Ken Hanlon, made a pointed observation about learning only by ear, especially when jazz students expressed frustration about studying classical theory. He said, "If you are only playing by ear, then you run a much higher risk of being derivative." He's right. But note, he did not say, ". . . you will be derivative."

My studies with Roscoe Mitchell deepened the value I place on practicing different musical disciplines. I do not consider myself an improviser, but I do perform using improvisational skills, and they absolutely influence my compositional output. In Roscoe's mind, there was no distinction between improvisation and composition. And when listening to him improvise, one can clearly hear why.

While we were in Iceland working with the symphony, he gave a solo concert, performing several improvised pieces in succession. One of these improvisations, for soprano saxophone, left the audience asking afterward, "How long had he been working on that piece?"

Daniel McKemie
October 2019
Berlin, DE

For a Computer Controlled Synthesizer

D/A /// A/D Liner Notes

D/A /// A/D is an album comprised of nine pieces released in 2020, exploring the computer as a hardware controller and the highlights of several digital audio applications and custom programs. The record is available to listen online and on CD, with artwork by Eri King and Daniel Greer (Eridan)

This album is a curated series of works that illustrate a few years' worth of work on computer-controlled synthesizer. The earliest pieces of mine that explored this hybrid reality could largely be chalked up as fancy noise studies (at best). These were mostly attempts to understand both sides of the system respectively, let alone how they could possibly work together. I do not usually aim to discuss technical details in liner or program notes, but this is an area that I have spent an incredible amount of time researching and developing music in, and have plans to continue doing so for the foreseeable future. It is in my hopes that these liner notes will motivate others to explore this topic, or at least spark a conversation about it.

Without an entire history lesson in this idea, this came about as a simple interest in wanting to join the power of computers with the interface and dynamics of control voltage. By generating programmed voltages in software, routing them in any number of ways, or even sending back voltage to be read by the software to in turn make decisions on control voltage generation, I see a rich atmosphere for electronic music making; both in live performance and in composed (or dare I say...algorithmic!) settings. The original experiments in this were constructed from patches written in Max/MSP and hooked in a myriad of ways to semi-modular Eurorack instruments. This then moved into constructing breadboard circuits and homemade hardware systems.

What I quickly realized in this venture was that neither my programming, circuit building, or general knowledge of modular synthesis were enough to warrant anything of value. But yet I moved forward. After spending some considerable time with at least two of these three topics, some musical ideas began to take shape. What I sought to do was utilize a number of different languages and approaches, as well as hardware systems to see how many variations could be achieved.

Using C/C++, CSound, JavaScript, Max, and Pd, I sought to explore every angle of expansion of a modular synthesizer. Creating interfaces, mobile device controllers, automatic voltage generation, and interactive performance environments, it was not always the case that one piece of software was paired with one piece of hardware. These combinations were smashed all together, programs combined with different interfaces, the same programmed procedures realized in different languages to explore the differences, and a huge number of hardware variations were all at play. I looked to the pioneers of the fields of tape music, live electronic music, and computer music for inspiration. I aimed to program some of their techniques and bring them into my own work, not as theft but as tribute (but you can be the judge of that) in order to construct a new way of music making. The beauty of electronic music is that the technology that is used to execute this music is always at the forefront, but sometimes it is the classic tools that are most engaging and intriguing to use. Because of the rapid pace in which technology evolves, we sometimes forget that some tools even existed, or that some tools can be used in ways not thought of prior to their usage falling out of style.

I settled on the pieces presented here for two reasons, the first being that they are the most musically interesting to me, and the second being that they exhibited an array of different approaches with varying degrees of success. Some of these pieces are performed live, some recorded live as an automated musical process, and some constructed as fixed media from either of these two aforementioned methods. What was learned in the end, and what is almost always learned in the end, is that it is not the technical specifications that make the work, it is the person behind it who makes the aesthetic choices on how; to deal with this technology however it does not mean that discussing technique (be it technical or aesthetic) has no implied value.

This does not mark an end to this approach to music making for me, but rather the beginning of what I hope to be are a series of experiments. Additional work and research is being done by taking these ideas into the realm of live-coding, custom built instruments and circuitry, and developing software functionality for embedded systems at a lower level. In addition, I am continuing to codify these works with supplemental writing and research papers that I hope to have published in the future. As always, thank you for listening.

This article was written as a guest blog entry for TOPLAP.[3] *in 2020. At the time of its publication, piecing together all of the steps and requirements for interfacing a computer with a modular synthesizer was a challenge. The goal of the article was to clearly illustrate and document my approach—without relying on specialized hardware or audio interfaces. ChucK proved to be the most effective language for TOPLAP's mission, as it is both robust in its ability to generate continuous signals for control voltage and easy to script on the fly.*

Leveraging computing power to generate and process control signals in a modular synthesizer is a practice with a storied, yet often forgotten past. From the first musical realizations of the 1950s through the 1980s, computers could take days to generate only a few minutes of music. In the late 1960s, "hybrid systems" like Max Mathews' GROOVE and the Roland MC-8 Microcomposer used analog synthesizers to produce sound while relying on computers to calculate the musical events. With the introduction of digital FM synthesis in the Yamaha DX7 in 1983, these systems largely faded away.

Today, with the affordability and renewed popularity of analog modular synthesizers, alongside the rise of computer music performance

[3] `toplap.org`

through live coding, linking these two worlds offers a compelling framework for live music. The writings presented here explore the benefits of using ChucK as a live coding tool for generating and processing control signals, dynamically altering and distributing audio effects across the synthesizer, and using the computer to map and monitor signal inputs—creating evolving, interactive performance environments in real time.

Intro

The purpose of this article is to encourage more musicians to explore a hybrid system of live coding/computer software and modular synthesizer, and to expand the capabilities of each by using the other. It is the opinion of the author, based on personal experience, that standing against certain musical or computational tools for extramusical reasons (i.e., a purist approach) is unnecessarily prohibitive. The author also believes that artists should, first and foremost, work with the means they find most effective and inspiring, while remaining open to changes in philosophy and outlook as they arise.

History

Early computer music works often took hours or even days to generate a small amount of music. It wasn't until the 1980s that personal computers became affordable and powerful enough to support complex synthesis. Hybrid systems, in which the computer handled compositional and algorithmic functions while the analog synthesizer produced the resulting audio, represented a significant leap in compositional productivity.

This began with Max Mathews' GROOVE system in 1969, which monitored performer actions on a voltage-controlled synthesizer. These actions could then be edited, recalled, and saved on a computer for future use. The Roland MC-8 was another important development: a commercially available computer paired with a CV and gate output interface. Notes and events could be programmed into the computer and routed through the interface to a modular synthesizer. This device, in concept, closely resembles the kind of setup this article seeks to explore.

Hardware Clarifications

For the sake of clarity, this article defines the difference between audio and control signals as follows: audio signals deal with sound, while control signals deal with structure. This closely aligns with Don Buchla's philosophy, whose instruments mechanically separated the two. However, most modular synthesizers allow audio and control signals to be used interchangeably.

Control signals are typically low-frequency signals, triggers, and gates used to modulate parameters or initiate events. Audio signals, on the other hand, are meant to be heard and are generated by oscillators and shaped by

VCAs, envelopes, and filters. That said, there is significant crossover, and these distinctions can become quite blurry, so readers are encouraged not to worry too much about strict definitions.

To achieve maximum versatility in low-level signal processing and generation, a few hardware components are necessary. First, you'll need an audio interface with DC-coupled outputs. This allows for the production of steady or slow-moving voltages that can be sent to the synthesizer. To prevent damage to your interface, it's important to use floating ring cables with a TRS connector at the interface and a TS connector at the synthesizer. Without these, one pole of the signal may have nowhere to go, causing the ring to short to ground. While the setup might still function without proper cables, incorrect wiring can damage your interface.

Alternatively, you could use TRS-to-dual-TS cables, sending the positive polarity on one and the negative on the other. There are several viable options here, depending on your synthesizer and interface. With minor cable modifications or a custom-built breakout box, you can achieve the same functionality.

For this use case, I'll be using a MOTU Ultralite Mk4 interface with floating ring cables purchased from Expert Sleepers. If you're unable to generate control signals in software, audio signals can also yield compelling results. As mentioned earlier, CV and audio signals are often interchangeable!

Live Coding

There are many options available for live coding and traditional programming languages, each with its own benefits and drawbacks. Having explored Sonic Pi, Gibber, TidalCycles, SuperCollider, FoxDot, and ChucK, I found that ChucK used with the miniAudicle IDE is the most preferred platform for interacting with modular synthesizers. It should be reiterated that personal preference is the primary driver in language choice here, especially with these more subjective comparisons. While similar results can be achieved in most of these environments, ChucK stands out in one regard: multichannel output is especially easy to implement, requiring no additional setup (e.g., editing SuperCollider's startup file), aside from specifying preferences in the IDE.

It's important to keep in mind that analog circuitry is not exact, this includes both the synthesizer and the interface. Interfaces behave differently in how they generate and receive voltages, and these behaviors can change over time. There are no fixed standards for voltage behavior across synthesizers, though sticking with a single manufacturer can help ease some of the complexities—but even that is not a guarantee. What works with one codebase and one synthesizer may not (and likely will not) work the same way with another. It is the performer's responsibility to understand their setup and know what adjustments are necessary to achieve the desired results.

To demonstrate a simple use of sending a control voltage, the code below represents a waveform with gain control, two fundamental elements when creating control voltage signals. The gain controls the level (or range) of the modulation being sent. In this example, we generate a pulse wave oscillating between minimum and maximum every 500ms, sent at a gain level of 0.1 (on a scale from 0 to 1).

```
// Osc to Gain Output (Left Channel)
PulseOsc p => Gain g => dac.chan(0);

p.freq(0.5); // Freq of output event
0.1 => g.gain; // Amp of voltage out

// Go for 1 day starting now
while(true) {
    1::day => now;
}
```

This output is plugged into the 1V/oct input of the VCO on the modular synthesizer. With the pulse width set to a 50% duty cycle, we get a change in pitch twice per second, once at the minimum, once at the maximum.

For clarity: a gain of 0 corresponds to a 2-volt output, and a gain of 0.1 corresponds to a 3-volt output. If the VCO frequency at 2 volts is 440Hz (A4) and at 3 volts is 880Hz (A5), then we would hear an octave shift occurring twice per second. If we adjust the frequency knob on the VCO to lower the pitch by one octave, the resulting output would fluctuate between 220Hz (A3) at minimum and 440Hz (A4) at maximum.

```
// Osc changed to ramp generator
SawOsc p => Gain g => dac.chan(0);

p.freq(0.1); // 0-1 over 10 sec
0.1 => g.gain;

// Go for 1 day starting now
while(true) {
    1::day => now;
}
```

The change in waveform introduces a gradual modification to the analog signal over time. With the same patching as before, with this signal plugged into the 1V/oct input of the VCO and the base frequency set to 440Hz, we start at 440Hz, glide up to 880Hz over the course of 10 seconds, and then suddenly drop back to 440Hz. The same gain adjustments still apply, but now the movement alternates between gradual and sudden.

If the frequency of the signal in ChucK were changed from 0.1 to -0.1, the reverse would occur: starting at 880Hz, gliding down to 440Hz, then snapping back up to 880Hz. This signal could be routed to any number of patch points on a modular synthesizer, such as the linear or exponential frequency inputs of the VCO, the cutoff frequency of a filter, the amplitude level of a VCA, a mixer input to combine with other signals, and so on.

An incredible luxury afforded to the performer when using this method of control is that connections can be reassigned without physically repatching the synthesizer. It functions like a digital pin matrix, allowing the performer to reroute and mix signals in real time with no audio dropouts, as shreds in ChucK can be swapped in and out seamlessly.

Audio Signal Control

One technique to consider is using recorded samples as a control source. It is the opinion of the author that blending both purely electronic and recorded sounds often leads to the most musically interesting results. There is no shortage of sampling and audio manipulation techniques available on the computer, though these fall beyond the scope of this writing.

To demonstrate, we use a recorded sample from Lou Harrison's instrument collection. The code below plays back the clip and allows the user to control both playback speed and gain for each iteration. The result is a blend of a sonically rich acoustic sound source changing over time and containing a wide range of harmonic content, with the analog circuitry of the synthesizer. This technique can be extended further through granular synthesis, a topic reserved for future discussion.

The code blocks below represent a simple piece linking several shreds in ChucK to a Kilpatrick Phenol desktop modular synthesizer. The synthesizer is equipped with two VCOs, two filters (low-pass and high-pass), two VCAs, and other utilities. It's important to note that on this instrument, all voltages are **bipolar**, except for gates. This is not typical for many modular synthesizer designs, which is why negative values are used in this code to access the full voltage range of the circuit. Your synthesizer may behave differently, so—as always—experimentation is highly encouraged.

I will use four shreds of code, outlined below, to perform this etude.

```
// Shred 1
SinOsc s => Envelope e =>
    Gain g => dac.chan(2);
10 => s.freq;
10 => g.gain

fun void osc (int freqMin, int freqMax,
             int long, int start,
             int length) {
    while (true) {
    Math.random2(freqMin, freqMax) => s.freq;
    Math.random2f(0, long)::ms =>
        dur t => e.duration;
    e.keyOn();
    Math.random2f(0, start)::ms => dur;
    e.keyOff();
    Math.random2f(0, length)::ms => dur;
    }
}
osc(-1, 10, 500, 1000, 100);

************

// Shred 2
// Randomly generated ramps changing
// in frequency every 100ms
SawOsc s => Gain g => dac.chan(3);
1 => g.gain;
```

```
fun void osc(int freqMin,
            int freqMax, int t) {
    while (true) {
        Math.random2(freqMin, freqMax)
            => s.freq;
        t::ms => now;
    }
}
osc(-1, -30, 2000);
************

// Shred 3
me.sourceDir() + "/../wavfile1.wav"
    => string filename;

if(me.args()) {
    me.arg(0) => filename
}
// The patch
SndBuf buf => Gain g => dac.chan(4);
10 => g.gain;
// Load the file
filename => buf.read;

// Time loop
while(true) {
    0 => buf.pos;
    Math.random2f(.2, .5) => buf.gain;
    Math.random2f(1, 10) => buf.rate;
    1000::ms => now;
}
************

// Shred 4
me.sourceDir() + "/../wavfile2.wav"
    => string filename;
if(me.args()) {
    me.arg(0) => filename
}
```

```
// The patch
SndBuf buf => Gain g => dac.chan(5);
10 => g.gain;
// Load the file
filename => buf.read;

// Time loop
while(true) {
    0 => buf.pos;
    Math.random2f(.2, .5) => buf.gain;
    Math.random2f(.5, 5.5) => buf.rate;
    1000::ms => now;
}
```

Conclusions/Further Work

A natural next step on this path is to reverse the relationship, sending signals out of the synthesizer and into the computer. Processing CV signals from hardware opens up powerful utility tools that can be programmed on the computer, enabling even more customizable modularity within performance systems. Mapping the modular synthesizer in this way allows for the construction of interactive environments that can read subsonic structural control voltages and make decisions based on their shapes and behaviors. And of course, audio processing and effects can also be added to enhance the depth and richness of analog circuitry. Simply put, all of the techniques outlined in this article can be explored in reverse.

Using TouchOSC enables the creation of custom interfaces to interact with the computer, the synthesizer, or both. Going beyond the standard knobs and sliders typical in modular systems, TouchOSC supports XY controls, multitouch grids, signal routing maps, and trigger-based preset launching. Because OSC can send and receive floating-point data, it pairs exceptionally well with CV generation and processing.

It is the author's hope that this article has opened up new possibilities for curious readers interested in incorporating modular synthesizers into their live coding setups and vice versa. Expanding both toolsets can only serve to deepen and enrich the musical experience.

Daniel McKemie
August 2019
Brooklyn, NY

DIY Piece 7

for Larry Polansky

For four (or multiples of four) voices

Within one canon (round or not), all of the voices must be capable of moving or operating in similar ways. This does not mean all voices must be homogenous. As you read on, this will become clearer.

Some type of trajectory, or constant line on X/Y axes is drawn on a piece of paper or realized in some other fashion. It should be a single line that always moves forward and cannot backtrack on either axis.

Once drawn, this line's x axis is assigned a given a length of time or some other constant rate of change. The line's y-axis is assigned a range to be used for ONE of any number of musical parameters: pitch, volume, density, thickness, activity, or any other constant the realizer chooses.

There can be ONLY ONE parameter assigned to the Y-axis and this parameter is uniform for all voices, the rest of the musical parameters are left to the performers. This is further explained later. The range's lowest point is the bottom of the Y-axis, while the highest is at the top.

The time/rate qualities can be chosen by the realizer or determined at random. Whichever quality is chosen, its divisions must be equal, so that has even proportions (i.e., the first half is the same length as the second, and so on) from beginning to end.

Create a Four-Voice Canon

X-axis

This initial trajectory in terms of time/rate (X-axis) is assigned to Voice 1 with the ratio of 1:1; with the remaining voices being contracted by the ratios of 5:7, 3:5, and 4:9, respectively. In this example, we will say that the X-axis totals 100 seconds. The voices will be of these lengths:

- Voice 1 - 1:1 = 100 seconds
- Voice 2 - 5:7 = 71 seconds (500/700)
- Voice 3 - 3:5 = 60 seconds (300/500)
- Voice 4 - 4:9 = 44 seconds (400/900)

These voices are to enter at the appropriate time so that they all end simultaneously. Rounding numbers is okay. This line can also be some variable of rate and is not constricted to the length of the voice only.

Y-axis

The musical parameter assigned to the range is left to the realizer. Again, only one parameter may be assigned and ALL voices share the same parameter throughout. For example, if volume is chosen, the bottom of the Y-axis is the quietest possible volume, the top being the loudest. These extremes of the Y-axis hold for every parameter given the instrument(s) it is played. No matter which parameter is chosen, the extremes on the particular voice's GIVEN INSTRUMENT are to be honored individually. Not all voices need to operate in extremes, but they must operate fairly within the same boundaries of each other

All other parameters are left to the discretion of the realizer or the performers so long as they do not eclipse or interfere with base parameter that the canon is exploring.

Four Four-Voice Canons

The piece can exist with the instructions given so far, but ideally one should create four of these four-voice canons and combine them together into one larger canon. While their voices should all be baed on the same line, each of the four canons can have their own y-axis parameter and share the same time on the X-axis. An example of how to realize this (using placeholders) follows:

- Canon 1 - Y = pitch
- Canon 2 - Y = density
- Canon 3 - Y = amplitude/dynamics
- Canon 4 - Y = ‘noisiness’ of timbre

Each of these canons is realized with the same voices being assigned the same positions/ratios. These parts are taken and then expanded according to the following ratios (with the same values as before):

- Canon 1 - 1:1 = 100 seconds (100/100)
- Canon 2 - 9:8 = 112 seconds (900/800)
- Canon 3 - 7:5 = 140 seconds (700/500)
- Canon 4 - 3:2 = 150 seconds (300/200)

These recordings are to be electronically manipulated to adjust playback speed according to these ratios, without changing the pitch. All of the recordings, if not round canons, should be placed to end together. It is encouraged to mix live performance alongside recorded parts.

DIY Piece 8

for Christina Stanley

For any number of voices

The realizer is to organize the score on a large canvas. The canvas consists of <u>objects</u> and <u>colors</u>. The objects are:

- lines
- triangles
- circles
- rectangles
- perfect squares
- pentagons
- hexagons
- heptagons
- octagons

These objects are created, copied, and cut from paper (or card-stock), to be placed on the canvas. These objects are of varying sizes, determined by the realizer.

The number of each object is chosen at random, between a minimum and maximum number chosen by the realizer (or also randomly chosen). The number/range and sizes of the objects should be proportionate to the size of the canvas.

After the objects are cut, they are combined together, gathered, and dropped onto the canvas in no planned way whatsoever. Objects that fall off of the canvas are collected then either discarded or dropped again in similar fashion. The objects are then organized on the canvas so as there are no overlays or collisions. If there is not enough space permitted to allow every object an independent space, the intruding object(s) should be discarded; however minimal overlays are allowed if so desired. Overlays should only occur if an artistic inclination calls for such an event.

Each object is then pasted to the canvas. They are then assigned colors in this way:

Seven colors are chosen, preferably a wide-ranging group such as red, orange, yellow, green, blue, indigo, and violet. The realizer may choose other colors so long as there are seven total colors chosen. Each color is assigned a number.

For example:

- 1 = red
- 2 = orange
- 3 = yellow
- 4 = blue
- 5 = purple

and so on. . .

Starting from the object at the top left of the canvas, a number is chosen at random, and that immediate object is painted fully, according to the chosen number. The next object to the right is then painted according to the same process. This is repeated per object per color, as if reading a book (top left to bottom right); until all objects are painted.

1) A navigation system on how to move through the score
2) Actions assigned to the objects/shapes

3) Actions assigned to the colors

4) Actions assigned to the empty space

Any other instructions not left determined by the realizer, are left exclusively to the performers.

DIY Piece 11

for Melody Loveless

For two pitched voices
Tempo = Slow (the same for each voice)

The piece is in Four Parts plus a Coda.

The steps to realize this piece are repetitive in many ways but vary from Part to Part. For organizational reasons, the order of operations may seem exhausting, but given the detailed change in each part, it is the most clear. If you follow the steps exactly, the piece should take shape just fine.

Part 1

1. The duration is determined by the realizer, either by choice or at random. For Part 1 the unit determining the duration is a number of phrases between 19-37.

2. Phrase length is determined by the number of eighth notes in each phrase, between 5-17. The length of each phrase is determined separately for each voice, either by choice or at random, therefore phrase lengths are staggered. Part 1 is made up of constant eighth notes. Naturally, due to phrase lengths being different for each voice, once EITHER voice reaches the number of phrases determined in Step 1, the Part should begin its conclusion. The realizer must decide to either augment or diminish the length of one voice to make them both align.

3. Choosing each pitch is a repetitive process. For each voice, the realizer must determine the initial pitch of the phrase. The initial pitch is chosen at random in Part 1 to be C, G, F, Bb, Eb, or Ab, and is placed ON the respected staff.

4. The resulting pitches are then determined based off of the previous pitch for the rest of the phrase. To determine the next pitch, choose at random:

A. If the next pitch will be higher or lower than the previous pitch. (see APPENDIX before continuing)

B. The interval. In Part 1 the choices are: P0, M3, P4, P5, m6, P8.

Reference the appendix at the end of this score as details are given for resolving issues regarding realizing these pitches, including how to solve "out of range of instrument" issues.

5. Once the phrase is finished, repeat Steps 3 & 4 until the phrase length is reached.

Part 2

1. The part's duration is determined by the realizer, either by choice or at random. For Part 2 the unit determining the duration is a number of phrases between 23-41.

2. Phrase length is determined by the number of eighth notes in each phrase, between 7-19. The length of each phrase is determined separately for each voice, either by choice or at random, therefore phrase lengths are staggered. Part 2 is made up of constant eighth notes, with the LAST 2 beats of every phrase being a half note. Naturally, due to phrase lengths being different for each voice, once EITHER voice reaches the number of phrases determined in Step 1, the Part should begin its conclusion. The realizer must decide to either augment or diminish

the length of one voice, to make them both align.

3. Choosing each pitch is a repetitive process. For each voice, the realizer must determine the initial pitch of the phrase. The initial pitch is chosen at random in Part 2 to be C, G, F, Bb, Eb, Ab, Db, Gb, B, E, A, or D and is placed ON the respected staff.

4. The resulting pitches are then determined based off of the previous pitch for the rest of the phrase. To determine the next pitch, choose at random:

A. If the next pitch will be higher or lower than the previous pitch.

B. The interval. In Part 2 the choices are: P0, m3, P4, P5, M6, P8.

5. Once the phrase is finished, repeat Steps 3 & 4 until the phrase length is reached.

Part 3

1. The part's duration is determined by the realizer, either by choice or at random. The unit determining the duration is a number of phrases between 17-45.
2. Phrase length is determined by the number of eighth notes in each phrase,

between 7-19. The length of each phrase is determined for each voice together, either by choice or at random, therefore all phrases will line up perfectly. Part 3 is made up of constant eighth notes, with the LAST 2 beats of every phrase being a half note. Unlike previous parts, the staggered problem at the end of the Part does not come into play as each voice's phrases are always the same length.

3. Choosing each pitch is a repetitive process. For each voice, the realizer must determine the initial pitch of the phrase. The initial pitch is chosen at random in Part 3 to be Db, Gb, B, E, A, or D and is placed ON the respected staff.

4. The resulting pitches are then determined based off of the previous pitch for the rest of the phrase. To determine the next pitch, choose at random:

A. If the next pitch will be higher or lower than the previous pitch.

B. The interval. In Part 3 the choices are: P0, M3, P4, P5, m6, P8.

5. Once the phrase is finished, repeat Steps 3 & 4 until the phrase length is reached.

Part 4

1. The duration is determined by the realizer, either by choice or at random. For Part 4 the unit determining the duration is a number of beats between 71-114 beats.

2. Part 4 is made up of dotted-quarter, half, dotted-half, and whole notes. The voices are again rhythmically staggered. For each voice, determine the note value from the options above, one-by-one until the number of beats is reached. To align the voices into the next part (as before), make the proper decision to diminish or augment a voice.

3. Choosing each pitch is a repetitive process. For each voice, the realizer must determine the initial pitch of the phrase. The initial pitch is chosen at random in Part 4 to be C, F, Eb, Db, B, A and is placed ON the respected staff.

4. The resulting pitches are then determined based off of the previous pitch for the rest of the phrase. To determine the next pitch, choose at random:

A. If the next pitch will be higher or lower than the previous pitch.

B. The interval. In Part 4 the choices

are: P0, m2, M2, m3, M3, P4, P5, m6, M6, m7, M7, P8.

5. Once the phrase is finished, repeat Steps 3 & 4 until the phrase length is reached.

Coda

1. The coda is 17 notes long for both voices.

2. The Coda is made up of dotted-quarter, half, dotted-half, and whole notes. The voices are back in rhythmic unison. Determine the note value from the options above, one-by-one, at random, until the number of beats is reached.

3. Choosing each pitch is a repetitive process. For each voice, the realizer must determine the initial pitch of the Coda. The initial pitch is chosen at random from ANY pitch, and is only done once at the beginning, again for each voice.

4. The resulting pitches are then determined based off of the previous pitch for the rest of the Coda. To determine the next pitch, choose at random:

A. If the next pitch will be higher or lower than the previous pitch.

B. The interval. In Part 4 the choices are: P0, M2, M3, P4, P5, m6, m7, P8.

5. Repeat Steps 2, 3, and 4 until the end of the piece.

6. For the Coda, the tempo is molto rubato, and each player attacks their notes at the same time, at their own determined pace. The piece is over when the last note is struck.

<u>Appendix:</u>

- If a pitch is determined to be out of range on an instrument through any process, then the direction of decided movement from the previous pitch, is reversed. The interval is not inverted, but merely goes up instead of down, or down instead of up, whatever the case may be.

- When placing the initial pitch on the staff, if there are two options, choose whichever you like, or at random.

- Dynamics are left to the performers, along with any relevant phrase structuring, accents, rubato, and so on.

- Any wind instruments should demonstrate circular breathing or make/take breath(marks) that do not interfere or cause large breaks.

- Realizers may change the tuning system slightly, given that the octave is still split in 12, and intervals are not crowded on one end of the octave.

- Instruments capable of playing multiple voices (keyboards, etc.), may realize an additional voice, so the player plays in two hands. So, for example, piano and violin decide to realize a score, the piece could technically reach three voices. With that said, four is the maximum number of voices that should be realized.

Maximalism

Maximalism is a series of recordings and essays that deal with the multidimensional properties of recorded sound within the context of modern compositional aesthetics. It is the author's attempt to make sense of the current state of things, to speculate upon where we may be heading next, and to consider why this aesthetic direction may be of some inherent value.

Time in music has been well-considered and explored, and like all things experiential, is necessarily time-based. Still, music as a time-based art often prioritizes the importance of the **x-axis** (ie. the past, present and future of a musical work), when at minimum two other planes exist.

In keeping with the Cartesian model, the **y-axis** generally represents the frequency domain while the **z-axis** can express timbral depth. Columns are constructed across the spectral range, and layers are weaved together in order to mask, support, and contrast the others on top of and behind each other. If the **x-axis** is not prioritized or even considered at all, the evolution of a piece can then be focused on these other axes and their compelling relationship.

While the **y-axis** represents frequency, in this context it is not treated the same as pitch. Pitch implies a relation to other pitches, generally located at fixed positions, and often in some functional relationship with what precedes and follows. Yet this is not in defense of supposedly non-functional pitch relationships nor microtonal possibilities. Rather, this approach to the **y-axis** is meant to highlight that the location within the frequency range in which a sound exists is directly tied to its timbre. For instance, the lowest note of a violin sounds much different than its highest. A raised awareness, if not prioritization, of an instrument's spectral range can assist in the production of sounds that occupy a unique place in a musical work to exist on their own, as opposed to being relegated to filler voices, passing tones, functional harmony, and other *support* roles. Additionally, having an understanding of the spectra an instrument is capable of can allow for greater coloration when met with other instruments or electronics.

The **z-axis** can be approached in a similar vein as space. It is not a matter of texture, but rather, our perception of the depth or *physicality* of sound. When a single pulse is traveling from left to right and back in our earphones, what is really happening? Is the sound traveling through space? No. It is simply a coordinated and opposing change in amplitude in each respective speaker in such a way that it is *perceived* to be traveling through space. *Depth construction* is an aesthetic technique, not a mechanical one, despite the use of devices of a mechanical nature that enhance one's ability to perceive depth. Since we are experiencing time no matter what, there is no need to focus on it even more than already needed. We can reconsider what value it brings to one's perception of music. The value placed on measured time is a compositional choice, but as listeners we can adjust our focus toward certain sounds in our physical space, with or without reference to the temporal relationships created by the performer. This happens all the time as performers, we shift our listening focus so we can stay synchronized with other players (especially in larger groups), while also considering how one's sonic contribution impacts the overall *depth*. This approach can just as well be applied to a composer's practice of compositional aesthetic.

By shifting one's focus toward the prioritization of the **y- and z-axes** over the **x-axis**, one emphasizes an exploration *into* sounds versus of sounds. Be they acoustic or electronic. **y** influences **z** and vice versa, while **x** operates either independently or dependently of the others.

Multiple instances of these grouped axes exist in a dimension all its own, a fourth axis such as that of a tesseract; where frequency, timbre, and time take on new meanings and assignments with each iteration.

Engaging with multiple instances of grouped axes, or sound blocks, can be likened to that of gazing into an aquarium. The tank of water itself has an **x-, y-,** and **z-axis**, and within these boundaries is a space in which many other objects may or may not exist, each having their own physical properties. For instance, picture a cube floating within the aquarium tank. That cube has its own respective **x-, y-,** and **z-axes**, and in this context, could be considered as a representation of some sound within space-time, with its own length, frequency spectrum, and timbral characteristics denoted by the properties of its shape. This shape is then located within a veritable universe (the tank) of like and unlike objects that exist in similar fashion.

A cube can be present in the tank and drift behind a much larger cube in the space, a masking of sorts, yet both cubes still exist, although one may have a more magnified presence. The cubes may also vary in size over time, representing changes to a sound's respective duration, frequency range, and timbral depth. These axes are not fixed and can change over time as they drift throughout the space (the fourth dimension). A piece of music that is fixed in some medium, a recording for example, is said to be the same every time it is played. Listening environment and amplification technologies aside, this is likely the case. However, slowing down the playback speed by half will immediately produce a different compositional reality while retaining certain identifiable characteristics as the *original*, just as Brahms' *Piano Concerto No. 1* can be played at half tempo. The composition is the same, but the cubes, as it were, have been relocated within the dimensional axes of the aquarium. If the intent is instead to simply change the dimensions of the sound through brute force means such as a change to playback speed, then the composition remains wholly intact. The cubes remain in intact, and while their appearance may have changed, these changes are consistent, relatively speaking, to every other cube.

The perception of time is a widely studied field across multiple disciplines. Time can be considered as absolute, relative, observable, perceived, and any variety of permutations of these elements. It is impossible for one person to observe time in the exact same way as any other and the perception of passing time increases with age. This is plainly seen in the classroom, workplace, and concert hall on a regular basis; activities that may seem boring cause us to perceive the passing of time as much slower than those that are engaging and exciting.

Performance is the creation of music in time, composition is the creation of music in space, and sound art is the suspension or removal of musical time. While the time element for all three can, and often are, represented by the **x-axis**, each one is treated differently. Performance, be it live or recorded, is the complement to composition, a physical and time-based act required to bring the piece into existence. The composition itself may exist as a score, graphics, theories, philosophies, or the media itself in which it exists (ie. the magnetic tape), but they all still have a performative complement whether it be potential or kinetic.

I have little to no interest in upending the entire western musical system or the laws of physics, but I do see value in considering the perception of the performative complement of musical compositions. Is it required, or even necessary, that the listener experience a piece of music as the composer intended, regardless of its existence as a written score, recording, or something else? Clearly the answer is no, otherwise the orchestras of the world could have stopped their incessant *reimaginings* of the classics after the first recording of Beethoven's *Symphony No. 5* was made available. Each recording and/or performance brings new and interesting perspectives, and if performers and conductors are afforded these interpretive liberties with how, and how often, compositions are presented, listeners should be afforded, at minimum, those very same liberties.

Historically, a composer writes a piece of music, a performer plays it, and an audience listens. A composition belongs to the composer and once it is handed over to the performer(s), ownership is spread amongst all involved members; the performer(s) play the composition and the audience listens, gaining partial ownership of the experience.

Like time, timbre is an equally useful, and experientially diverse, candidate for defining musical form. One way is by pairing time and timbre together and using textural mechanisms to define the boundaries of a composition, much in the same way time and pitch are often paired. For the classically trained, it may be difficult to completely pull away from pitch structures when discussing musical form, but percussionists have been doing this for years. It should be noted that rhythm is in fact, time itself musically personified.

The exploration of *depth* along the **z-axis** is an exciting avenue when conceiving new forms and new modes of listening. This exploration of *depth* follows in the spirit of John Cage's consideration of space and perception: You are walking on the sidewalk, and you see your friend on the other side of the street, you wave to your friend and make eye contact, a truck drives down the road and obstructs the view of seeing your friend, does this mean that your friend is no longer there? Of course not. The **z-axis** can be treated the same way. The perception of sound does not need to be directly tied to its presence or the ability to hear it as a distinct entity within a larger collection of sounds.

It has been my intention to illustrate here that sound as it exists in this fourth dimension, including the **x-/y-/z-axes**, are not fixed but dynamic. This is the same mental space that one enters when pondering time, the universe, the infinite and infinitesimal, and multidimensional shapes, such as the tesseract, in a three-dimensional world.

Positioning the frequency domain along the **y-axis** is arguably the simplest and historically-embedded axis to determinately control. The pitch-centric foundation of Western notation lends itself perfectly to this concept. In electronic music, a single sound source as simple as a sine tone can be swept along this axis seamlessly and with a high degree of specificity. Traditional western music focuses on the use of predetermined gradations of the frequency spectrum, ie. or pitch, in a functional manner. Harmonic motion like *ii-V-I* and *I-IV-V* have worked for hundreds of years and are still effective in the structural development of tonal music. There are also alternative tuning systems informed by non-western musics and mathematical thought.

Twelve-tone and serial music democratized the pitch field, microtonal music divided the octave beyond the Western 12 pitches, spectral music used frequency analyses of timbre to inform pitch content, and electronic music represents a completely a breadth of sonic practices with no particular reverence.

Karlheinz Stockhausen applied serial techniques to electronic music, a significant move, but an even more important exploration was the use of reverb. Beginning in his earliest works *Studies I & II*, by using only sine tones and space he created an incredibly rich tapestry of frequencies using only the most focused timbral element. One striking feature of these works is the use of space to expand the **y-axis**. The focus was not so much about the frequency of the sine tone by itself on tape, but the combination of tones created when reverb was introduced. This is easily replicated with current technology by taking a test tone, routing it through any reverberation module, and increasing the amount of reverb; one begins to hear a growth of sum and difference tones relative to the effect. In the 1950s this was done using physical spaces and can now be done with even the simplest audio plugins or hardware processors. The details of a space, real or imaginary, can be defined to make even the simplest of sounds have a much more prominent presence in the frequency domain. This can be moved to the physical space using devices like spring and plate reverb units as well as multichannel playback using a variety of approaches to speaker placement. While space is potentially quite modular in this regard, the most modular piece of this particular sonic puzzle is the listener and their position in said space.

Refraction is the deflection and changes to speed of sound waves based on how it travels through and around mediums of varying density. This also applies to light, radio, and electromagnetic waves, and to this end, something like an aquarium showcases this phenomenon quite clearly. The angle at which the viewer peers into the tank can drastically change the general appearance of the objects inside and distort one's perception of their location in the three-dimensional space. As a sound in the tank is represented by a cube, the characteristics of that sound may change based on the position of the listener.

If any barriers exist between the sound source and the listener, selected frequencies may be filtered and/or amplified. If the listener moves to a different location in the space, the sound may reflect off of a different surface, enabling different areas of the spectra to be filtered and amplified. Depending on the sonic qualities of the sound source, pure feedback for example, simply turning one's head can cause quite drastic shifts in what frequencies are perceived by the listener.

The **y-axis** as a descriptor of some sonic entity can expand beyond pitch or frequency information and into the physical space. But how does one replicate or realize these spaces outside of the recording? These spatial characteristics could inform the sound material itself, be it frequency or timbre. A common approach is to process the audio using spatialization and reverb after the music has been written, effectively repositioning two-dimensional audio into a three-dimensional space. A proposed idea is to use multiple processing units for space versus one room that is defined at the end of the chain, and to plan these spaces *first* before constructing any audio material. In other words, writing *for* or *into* the space, albeit an imaginary space generated by software and/or hardware. This requires the development of a space, or spaces, and that although these spaces may change over time in some (in)determinate way, this *space* template of sorts will likely have an impact on the creation and realization of the audio content for the piece.

Daniel McKemie
March 2021
Brooklyn, NY

Theia Mania

Theia Mania is a three-movement work written between 2022 and 2024 that eventually became part of a retrospective of pieces dating back to 2008, all of which involve acoustic instruments or samples combined with electronics. These are the liner notes of that release, found on Modulisme.[4]

Coda (2024)

While seemingly out of place in the track order, *Coda* is the first work I wrote taking a new approach of slowing things down in my process while more deeply exploring the aesthetics and creative goals outlined in these notes. Having moved all my equipment back home from my studio, it is an area that forces me to do all performing, recording, sampling, and effects

[4] Modulisme (translates Modularism) is an imprint documenting left-field Modular Synthesis. A platform that aims to support original composing, for analog modular systems but not only...out of Marseille, France and run by Philippe Petit. `https://modular-station.com/modulisme/`

processing on analog hardware. The purpose of this is not around gear fetishization, but instead to really examine the fundamentals of effects processing, synthesis, mixing, editing and listening. It's also a love letter.

Lower Springs(2009)

This was one of the first works I composed as a graduate student at Mills College, in James Fei's Electronic Arts course. The class assignments were designed to explore topics in aesthetics, such as process art, sound art, mixed media, video, and film. *Lower Springs* was originally written for a video piece, which was eventually scrapped, leaving only the music. I found some inductor mics in the Prieto Lab, where the class was held, and began exploring the sounds of electromagnetic fields from devices around my studio. The technical setup was intentionally kept simple to focus on the audio itself, routing it through a mixer, recording onto 1/4" tape, then remixing, splicing, and bouncing onto a second tape machine, resulting in the final work.

Metals (2008)

While I had been writing, performing, and releasing music under different monikers for a few years by this point, this is my first serious composition. Originally conceived for percussion and tape, the concept revolves around the acoustic and electronic manipulation of resonant metals.

The piece was recorded, processed, and mixed on a Fostex MR-8 Digital Multitrack Recorder—a rather tedious device with uninspiring effects. The acoustic elements are more prominently featured, as I was far more comfortable with these than with electronic ones. Cymbals, gongs, tam-tams, homemade devices, and crotales were bowed, struck, and rolled; submerged in water; taped in parts to isolate harmonics; played against an open piano; on timpani with the pedal bending the pitch, and more. The goal was to capture all these sounds with the portable recorder and build a tape part to accompany a live score, but as the music reached a certain level of completeness, it remained solely as a fixed media work.

Decontrol (2020/2022)

Originally written for live-stream performances during the COVID lockdown, I adapted this piece for the concert hall at the 2022 New York City Electroacoustic Music Festival. The audience is invited to visit a web page that allows them to directly control the soloist's instrument. This page features an interface with sliders and buttons that transmits high-resolution, 14-bit MIDI data via WebSockets to the performer's computer. The data is then converted into control voltage signals, which are sent to various points in a modular synthesizer and other handmade circuitry, effectively inviting the audience on stage to turn the knobs themselves. The performer has no prior knowledge of how the audience will interact with the web page, what information it will transmit, or when the transmission will occur.

Zen in the Spectral Key of A (2012)

This was my first attempt at combining acoustic and electronic elements in a notated composition. Heavily inspired by the work of James Tenney, I aimed to create a space where electronics could interact with an instrument without any live processing. This idea emerged when Ryan Ross Smith and I were recording several of his animated scores in upstate New York. After a break in the day's work, I shared this score, and we discussed ways to bring the piece to life. There was a racked Moog Voyager, without a keyboard, and we proceeded with rehearsing and recording. For me, the significance of Zen was that it successfully conveyed the notated work for live electronics I had envisioned, with minimal discussion and planning needed from the performers. Over the years, various approaches to live electronics notation have been used, from listening scores to texts to graphics. I remain deeply interested in exploring the tension between the openness of electronic instruments and software and a system of musical notation to achieve compositional results while preserving the performer's voice. The work in Zen laid the groundwork for my radio opera Pneuma, which premiered at Wave Farm in 2014.

Theia Mania Mvts II & III – Floating Feet (2023) / *Slaydie's Knees – Part Infinity* (2024)

Theia Mania is a three-movement collection written between 2022 and 2024, consisting of *Sample Systems*, *Floating Feet*, and *Slaydie's Knees - Part Infinity.* Though not the only music I composed during this time, these works share a common focus on aesthetics, with no technical goals or experiments in mind. After nearly six years of immersing myself in Web Audio, I needed to set aside technical aspirations and reconnect with my foundational musical instincts.

I don't typically share my personal feelings explicitly through art or attach empty meanings for the sake of "programmatic music." However, in this instance, these pieces reflect a profoundly emotional period in my life. Throughout the conception, creation, recording, and finalizing of these works, my wife Sarah was battling cancer, to which she ultimately succumbed to in the spring of 2024. Music was my sanctuary. Theia Mania embodies what the Greeks called "divine madness", the sensation one might feel experiencing love at first sight.

Slaydie's Knees isn't a sequel but a tribute to earlier works created when I first met her. This series blended MRI recordings with synthesizers in a call-and-response format. Her suggestion to listen to MRI machines came from scans on her injured leg from roller derby, unaware at the time that such machines would later become mainstays in her cancer treatments. The piece doesn't include recordings or samples from these machines, as there are no more scans to be had.

This record is wholly dedicated to Sarah Marie McKemie, whose love and admiration transformed me in immeasurable ways. I will forever miss you.

Modulisme Interview

An excerpt from an interview as part of the release of Theia Mania.

Modulisme: *You are based in NY, how is the scene there? Would you tell us more about your surroundings? Composers you like, feel close to?*

Daniel: I've been in Brooklyn for nearly nine years, and I love it here. I always tell people that you can find at least a dozen others in the city who share any niche interest you can think of. There are several venues all over the city, some great underground and DIY spots, many well-run and professional ones, and they treat performers well. Others don't! But that's the case anywhere.

I haven't really latched onto any one scene or style. Since moving here, I've found myself attending more art shows and galleries than concerts. I love the freedom of moving through a space and discussing ideas and aesthetics with creative people working in different mediums as it's all so transferable. I do try to make it to concerts where I don't know anyone on the bill but have some idea of the music. It can be tricky, though, because venues here don't often specialize in a single type of music; the lineup can change completely from night to night, or even within the same day! A club might host a lecture meetup in the afternoon, punk bands at 7 p.m., and then transform into an all-night rave by midnight.

After doing this for so many years, you notice how people take different paths. Some move on to things outside of music, others experience meteoric rises and become incredibly famous, and many fall somewhere in between just working, practicing, and showing up day after day. I find that I'm most influenced by people I personally know or collaborate with, and there's no shortage of them in the city. The live coding scene is booming here, and while I don't participate in it, the DIY and community-driven ethic of that group is incredibly inspiring.

One person who stands out in that scene is Melody Loveless, whose music and teaching seamlessly integrate technology, performance, and aesthetics like no one else I know. Ryan Ross Smith, a dear friend and frequent collaborator, amazes me constantly with his staggering volume of musical output spanning multiple genres. Since diving back into academia, I've met more inspiring figures there as well. Luke DuBois always delivers an engaging and unpredictable show, flawlessly executed every time. And then there's Eri King and Daniel Greer, who work together as Eridan, blending visual and mixed-media art with music in a way that epitomizes creativity, no matter the medium.

Bringing it back to visual arts and galleries, this is why I'm drawn to those spaces. Practice is practice. Whether we're playing synths or violin, painting, or drawing, we're all honing our craft. Having been immersed in music for nearly 30 years in so many roles, I've developed an ear for recognizing the practice behind the finished piece. And that's what makes NYC's endless variety so thrilling is that the artistry here is a reflection of relentless, passionate work in every form.

Modulisme: *Your artistic practice covers many different mediums and aesthetics, and you were fortunate to study with Morton Subotnick in the fall of 2022. Would you develop on that? What did he teach you, and how did it influence the way you compose? What subjects have stayed with you and influenced your practice?*

Daniel: I was incredibly fortunate to study with Mort, and the latest pieces on this record are a result of new approaches picked up in these lessons. His lectures often centered around a core question: "If everyone has access to the same technology: software, hardware, or plugins—how do you use them to create something that reflects your unique voice?"

This is the same question one asks when writing for instruments, the same ones that we have had for centuries. The 20th Century saw an explosion of efforts to create new sounds, with a greater focus on timbre as a musical device. We can leave it to the musicologists to explore the details of why this shift occurred, but speaking to my own approach to composition over the last 20 years, it can be explained in a kind of hierarchical sense.

Music can be categorized by dimensions: pitch, timing (or rhythm), amplitude, timbre, and space. Historically, music theory and analysis focused on the dimensions of pitch and timing, with it being less common to focus on timing alone. In the 20th Century, when developments like the percussion ensemble (e.g., Varese) and the introduction of noise as a musical element emerged (e.g. Russolo, Avraamov, et al.). It is the reordering of this hierarchy, with timbre placed at the top, that introduces abstraction. If the public's focus is primarily on pitch and timing, it's no surprise that a shift in emphasis leads to debates over whether such music is even considered music at all.

Mort put it simply: "Music is what culture decides is music." Too much of my time as a musician working in the abstract has involved discussions ab out my music or music I enjoy, and whether it qualifies as music at all. It does. It is music because a significant part of our culture deems it so, and refusing to acknowledge another culture's appreciation of something is on the person holding that view. I no longer entertain these conversations with people who come from a judgmental place. There are too many important things to focus on to worry about such concerns! And I have little interest in conservatism in general. There is a culture around experimental and abstract music that has existed for well over a century, and it is an extension of the classical music canon, even for those who are making statements against it.

After pitch and rhythm, elements like form and orchestration are typically discussed in music. Orchestration marks the first real entry into timbre as a musical device, and it wasn't formally codified until 1844 with Hector Berlioz's *A Treatise upon Modern Instrumentation and Orchestration*, 100 years after the founding of the Mannheim School. While the orchestra has existed as a musical ensemble since around 1600, the modern version developed within the Mannheim School is arguably one of the most exciting musical tools ever employed. The capabilities of an orchestra are vast, with variable instrument groups covering nearly the entire audible range of human hearing, extended techniques, and the potential for spatialization or re-imagining (e.g., Henry Brant), and functions as a complete instrument.

Mort discussed the orchestra in depth, viewing the Buchla synthesizer as a template for building its instruments and then composing and arranging parts much like one would with an orchestra. He cites Earle Brown's *Available Forms* as a relevant piece to supplement the discussion of his own approach, though Mort had been working in this way for years before ever discovering the piece. In *Available Forms*, players are given parts that are initiated by the conductor, who provides further instructions on how to guide and cue their performance. While the details are too involved for this text, it is encouraged that any interested listener explore this piece further, as it was all in relation to find what Mort called, "a new new music".

One last guideline from his lessons is to maintain a policy of performing everything yourself. If you trigger or sample anything from the instrument, the sample should be something that you perform and record yourself with the same seriousness as you would when recording a classical piece in a studio: doing multiple takes, editing them, and ensuring the highest quality result. It is not enough to just sample, loop, and sequence without great care to the process of personalizing it.

I am incredibly grateful for the time spent with him discussing music and its history, technology, composition, the orchestra, and his many fascinating stores about American music and art and those who create it. Lessons that I will certainly carry for a lifetime.

Modulisme: *You like to break down barriers and find a dialogue between various (electroacoustic and instrumental) music cultures and heritages. Do you encounter problems related to preconceived ideas, limitations or rules imposed by these respective genres? How can you achieve such mixity?*

Daniel: I wouldn't say I find problems, or at least not ones that feel personal to me; but I deeply respect the canon, those who came before us, and those creating alongside us. I love and appreciate music of all kinds, but over the years, I've realized I'm most drawn to honest music. I enjoy analyzing music through the lens of styles, approaches, techniques, and when it comes to electronic music then also the equipment and technology involved. But in the end, what matters most is that the aesthetic and the statement feel honest. There's no formula for finding this honesty, but I recognize it when I hear it. It’s perhaps the magic of music that we all talk about and seek to understand. Mort might call it “qualia”—the indescribable essence that makes a piece of music resonate in a way nothing else can.

Modulisme: *What do you usually start with when composing?*

Daniel: I often spend a long time thinking about ideas and forms before ever sitting down to realize any of it. This is especially true with instrumental music. I’ll mentally craft a large portion of the ideas, quickly sketch them out on paper, then move into editing, workshopping, and rehearsing, if that’s an option, before engraving and finalizing the piece.

With electronic music, the workshop and rehearsal phases are always part of the process, so I dedicate significant time there. However, I try to stick to the initial constraints I set during the sketching phase. This approach has become increasingly fluid as my understanding of the lower levels of software and hardware has grown. The ability to refine musical ideas on a microlevel through deep technological manipulation has been incredibly exciting, even though it’s greatly expanded the time it takes me to complete a piece. Which I’m perfectly okay with.

Modulisme: *How do you see the relationship between sound and composition?*

Daniel: Since I first started writing music, timbre and form have always been at the heart of my work. As a classically trained percussionist and composer, I have a solid grounding in functional harmony, counterpoint, set theory, and other foundational concepts. While I've never been particularly interested in working with those elements explicitly, they've profoundly influenced the way I approach composition and sound.

I approach pitch as a timbral device rather than a functional one. For example, playing a low A versus an A three octaves higher on a violin produces drastically different timbres, particularly in relation to the surrounding sounds. In my music, in the vast majority of cases where I've written an A, you could replace it with an A# next to it, and the effect would likely remain unchanged. The pitch holds no traditional harmonic or melodic function; instead, its role is purely timbral, a choice made either by me or by the performer.

With all that said, I find that I'm most inspired by composers who work with the elements I don't typically use, because form is the unifying element of all music. Music is a time-based art, and whether you compose, improvise, use instruments, electronics, noise, or anything else, it all revolves around time. Form is what brings everything together. Philip Glass is one of my favorite composers, and though his sound is nearly the opposite of mine, his treatment of form and direction never fails to captivate me.

Sounds are the events within a form and the form is composition.

Modulisme: *Your compositional process is also based upon the use of acoustic instruments that you process or combine with Electronic. How do you work to marry that Electronic with your acoustic matiere?*

Daniel: The performative aspects of both instrumental and electronic music are completely intertwined for me, arguably inseparable, and this is where the marriage happens. A great example of this is in my work Pneuma, a multimovement, mobile-form piece for several soloists and fixed media. At no point in the piece is there live processing of instruments; instead, the performers play to a timeline of gestures alongside audio that represents an already processed version of their instrument. This showcases a relationship between two entities side by side, rather than following a chain like instrument → processing → end result. As I mentioned before, form is the unifying component of all music, and this approach allows for the creation of form in fixed media and along a timeline, where neither relies on the other. Instead, they function more like a duo. This method is consistent in all my music, whether for acoustic instruments, electronics, electroacoustic works, samples, or synthesis. I also like to keep the forms as modular as possible.

Modulisme: *Are you feeling close to some other contemporary Modularists? Which ones?*

Daniel: As I mentioned earlier, Luke DuBois is probably the most engaging player out there that I have seen lately. The blends of sonic and visual elements are striking and always well executed. Robert Aiki Aubrey Lowe, who is also here in New York, with his use of voice and clear command over the instrument, coupled with a great range of musical history that he has been a part of. Bob Ostertag for his music but also his commentary on the subject, especially in his book *Facebooking the Anthropocene in Raja Ampat*, he has a great essay on electronic music performance. Finally, I have to shout out Las Sucias, a Bay Area duo with Danishta Rivero and Alexandra Buschman, both fantastic musicians in their own right. Their live show is raw and powerful and incorporates highly creative uses of the instruments and ritual. I was fortunate enough to share a bill with them not long before moving to New York, and they have an album out on Ratskin Records that has been one of my favorites for years.

Modulisme: *Which pioneers in Modularism influenced you and why?*

Daniel: Pauline Oliveros was my first big influence that expanded what I thought possible in live electronic music performance. Her early works with tape delay feedback are incredibly groundbreaking and, in many ways, criminally overlooked. With essentially a few pieces of equipment, she creates an entire sound world, all in real-time, which was unprecedented at the time. She used that system for years, and her works with the Buchla and Moog with that are still among my favorites.

Mort's earlier Buchla pieces like *Silver Apples*, *Sidewinder*, and *Touch* are all a constant source of inspiration. I feel incredibly fortunate to have studied with both him and Pauline, discussing their compositional approaches and use of technology. It's really something to explore their music as a listener, gain an understanding, and then be able to sit and learn their techniques firsthand, feeling that understanding change. It's quite special.

Éliane Radigue's music is a favorite and demonstrates a level of focus, discipline, and control of the instrument that only a few can achieve. Her works prior to using the ARP 2500 have this too. Working in feedback is challenging for several reasons, and like Pauline, she maintains absolute control that really showcases a thought and practice you can clearly hear.

Composers who used modular systems for tape music have been hugely influential to me in the last decade or so. Bernard Parmegiani is arguably my top favorite in that category. His attention to sonic detail, form, and organization while maintaining an incredibly engaging 'liveness' in the music is unrivaled.

Modulisme: *Any advice you could share for those willing to start or develop their "Modulisme"?*

Daniel: Just do it! Experimentation is key. Take risks and always appreciate that limitation breeds innovation. Build, or learn to build, the modules if you can. It saves money and gives the instrument more 'soul.' Don't get hung up on equipment or the next best thing; work with what you have. It's always enough!

Epilogue

How should we treat technology in music? Who or what, is making the music? In traditional acoustic music, the instrument is the vessel from which sound is produced and projected, but it is almost always the performer who initiates and controls each sound. Automatic processes and conditionals cannot be implemented on a clarinet in the same way they can on a computer, synthesizer, or circuit board.

The composer can now create self-generating systems that unfold in real time, yet may also be performed and manipulated in parallel. This opens the door to an entirely new approach to music-making and once again invites the question: *Who* or *what*, exactly, is making the music?

Compositions must still be conceived, systems must still be designed, patches must still be built, and computers must still be programmed. Some technologies are more immediate and responsive for music-making, improvisation, and live performance, while others are better suited for executing detailed calculations and complex processes. Both analog and digital technologies are capable of supporting all these functions, each with its own strengths, weaknesses, and varying degrees of efficiency.

So, while a system can be designed, built, and executed to run indefinitely without user intervention, it is still the user who must design, build, and execute that system. It is still composing.

The performance of electronic music raises the point that one can implement automaticity and control its parameters over time while simultaneously performing within the same technological stack. Which is a stark contrast to acoustic music performance. In electronic music, the addition, removal, or substitution of technology can lead the user down previously untraveled paths. To craft a manifesto against a particular toolset, however, only serves as a hindrance.

The purposeful self-limitation of creative tools is an effective way for the user to better understand the technology they seek to use, and to gain a deeper understanding of their own relationship with those tools.

There are certain identifying structures within music that the public has come to expect when listening: melody, harmony, rhythm, guitar riffs, swing, groove, loops. There is a widespread tendency toward synchronicity, or at least toward something that "makes sense." Since 1945, however, a subset of composers has worked to expand the elements of music not included in that list: timbre, tuning systems, electronics, noise, silence. The modern composer operates in a radically different musical world, one that deliberately undermines the assumptions of traditional practice and challenges the expectations of the casual listener.

It is the role of the listener to shift their focus—to better align their presence with the context of whatever musical sphere the composer happens to be traversing. Edgard Varèse famously referred to music as "organized sound." While traditional composition certainly qualifies as such, the definition of what constitutes "organized" is always in flux, and it is up to the listener to follow that shift. Although Varèse wrote this over a century ago, debates still rage over what qualifies as "real music," suggesting a persistent stagnation in the listener's ability to navigate between evolving aesthetic contexts.

But this challenge is not limited to the casual listener. To experience sound for sound's sake is deeply liberating; yet even the most steadfast modernist, postmodernist, or experimentalist may struggle with it. This is due, in large part, to the enduring grip that our historical and cultural contexts continue to exert on us.

Looking at rock music, one of the most successful and recognizable genres in history, one of its defining characteristics is the guitar solo. Guitar solos, and the performers who deliver them, have captivated listeners for decades. In one of the most iconic solos of all time, Jimi Hendrix's rendition of the Star-Spangled Banner at Woodstock in 1969, he pushes the instrument into radical, noise-filled territory while abstracting the national anthem.

Beyond the cultural context of the performance, what makes it so special? It's the *sound* he draws from the guitar. Just let the sound be itself and let it take you on the journey.

Much of the public is put off by contemporary art and music because of the attitudes often associated with their creators and audiences. These attitudes frequently imply or even require, that in order to be understood, high art must be attached to a conceptual framework. Not unlike a joke that needs to be explained to be funny. When music or art requires explanation in order to be validated then it is a failure of the artist, not the viewer or listener. That said, explanations can absolutely enrich the experience provided by the original work.

It's fair for the public to be standoffish toward the arrogance often on display at new music concerts. But this attitude is not exclusive to the avant-garde, it is also present, in equal or even greater measure, within the more mainstream or commercial music worlds. Arrogance is not the domain of any one genre or practice; it's a broader cultural tendency that affects all corners of the arts.

For centuries, pitch reigned supreme in the hierarchy of musical elements. As evidence, consider that all salient aspects of a classical symphony, especially pitch and arguably to a lesser extent rhythm, can be reduced for performance by a single pianist. This emphasis on pitch has shaped the perceptions of both musicians and audiences for generations, creating a context that can be difficult to move beyond.

Music is a time-based art. One can stare at a painting indefinitely, but a piece of music has a beginning and an end. Rhythm is measured time in music. In works with very slow or nearly imperceptible changes, such as drone music, rhythm may not be perceived, even though it is always present. Rather than treating rhythm as merely the division of a beat or the presence of a constant pulse, it should be understood more broadly as a unit of time. Fear not if you cannot tap your foot! Because music, being grounded in time, also has the power to suspend it.

Daniel McKemie
January 2024
Brooklyn, NY

Meditation for your Guts

Sit completely still for several moments.
   Listen to your body.
      Your heartbeat.
         Your breathing.
            Your nervous system.
               Your blood flow.
                  Play.
               Listen to your bones.
            Your joints.
         Your teeth.
      Play.
   Listen to the sounds of your digestive system.
      Your swallowing.
         Your stomach pangs.
            Play.
               Repeat as desired.

Daniel Steffey
October 2009
Oakland, CA

List of Works

Electronic Works

- *Coda* (2024)
- *Floating Feet* (2024)
- *Slaydie's Knees - Part Infinity* (2024)
- *Sample Systems* (2023)
- *Practical Environments No.1 (BQE)* (2023)
- *Ototonia* (2023)
- *Wire Transfer* for two remote performers (2022)
- *Mass* (2022)
- *Decontrol for Live Electronics and Audience* (2021)
- *Covid Fridge at the Phoenix Hotel* (2021)
- *Maximalism* (2021)
- *Control Surfaces* (2019–20)
- *Digital / Analog /// Analog / Digital* (2016–2019)
- *Motions for the Tranzac No. 3* (2018)
- *Seven Movements* for modular synthesizer (2018)
- *Control Voltage Feedback Study No. 1* (2018)

- *Slaydie's Knees (Technically Speaking)* (2017)
- *Slaydie's Knees - Parts Nos. 1–15* (2017)
- *In Memoriam Pauline Oliveros* (2017)
- *Open Circuits* (2017)
- *Automatic Shortwaves* (2017)
- *For Wave Farm (Radio Synthesis No. 1)* (2017)
- *Interaction Studies Nos. 1–2* (2017)
- *Interactive Arduino Circuit No. 1* (2017)
- *Overlapping Frequency Circuit* (2017)
- *Meditation No.1* for Triangle and Electronics (2017)
- *Radio Synthesis No. 4* (2017)
- *Semi-Automatic Synthesis No. 4* (2017)
- *Semi-Automatic Synthesis No. 2* (2017)
- *Study for a Computer Guided Interactive Modular Synthesizer System No. 1* (2017)
- *Automatic Typing Nos. 1–6* (2016)
- *The Anatomy of Troubleshooting* (2016)
- *Motions for the Tranzac No. 2* (2016)
- *Phase Recognitions Nos. 1–3* (2016)
- *Automatic Synthesis Nos. 1–7* (2016)
- *For a Computer Controlled Synthesizer* (2016)
- *For a Synthetic Voice Nos. 1–3* (2016)
- *everyone always becomes yesterday's news* (2016)
- *Radio Piece No. 8* (2016)
- *Square Canons* (for Larry Polansky) (2016)
- *Variations on a Theme by Nick Wang* (2016)

- *Three Unfinished Movements for Fixed Media* (2016)
- *For Three Oscillators* (2015/16)
- *Motions for the Tranzac* (2015/16)
- *Pneuma* (2014)
- *Collage No. 2* (Yellow Dog) (2014)
- *justice/peace [null and void]* (2013)
- *Transistor/Multiband* (2011)
- *Dysrhythmia* (2011)
- *Discords Nos. 1–5* (2010/13)
- *(Micro)Cassette Music Volume 1* (2010)
- *oe* (2010)
- *(R*Sin)F=M* (2010)
- *Collapse* (2010)
- *Claqué* (2010)
- *+Sin(gliss)P* (2010)
- *Lower Springs* (2009)
- *Chloros* (2009)
- *Radio Pieces Nos. 1–7* (2009)
- *Sonal Mirror Whitewash* (2009)
- *Metals* (2008)

Instrumental Works

- *Cadenza for (Amplified) Solo Cello* (2011/2023)
- *Meditation for your Guts* (2009/2022)
- *SSD* for 1–12 performers (2022)
- *Seaswell* for four snare drummers (2020)
- *Composites 1* for soloist and four players (2019)
- *Journey [Back] to the East Bay* (2014)
- *...a completely rational idea of sound...* for percussion quartet (2013)
- *Cosmos/Anticosmos* for four glockenspiels (2012/13)
- *Transmissions for Pipe Organ* (2012)
- *New Peace* (2012)
- *Concerto for Electronics and 15 Musicians* (2012)
- *Double Duet for Two Cellos and Two ARP 2600s* (2012); co-composed with Benjamin Ethan Tinker
- *Dispersion* for sub-saxophone quartet (2012)
- *Surf Designs* for two violins (2011)
- *For Clarinet Solo and Percussion* (2011)
- *Concerto for Contrabass Clarinet and Nine Musicians* (2011)
- *Densities for Pipe Organ* (2010)
- *String Quartet No. 1* (2010)
- *No [...], I Won't* for contralto and vibraphone (2010)
- *Etude* for string quartet (2010)
- *Two Movements for Percussion Trio* (2010)
- *ae* for solo violin (2010)
- *Chloros II* for amplified violin and percussion (2009)

Pneuma (2014)

For 1–12 Musicians, Tape, Operatic Voice in Two Parts [for radio broadcast]

- *Pneuma I* – for vibraphone and tape
- *Pneuma II* – for bassoon and tape
- *Pneuma III* – for clarinet and tape
- *Pneuma IV* – for cello and tape
- *Pneuma V* – for electronics and tape
- *Pneuma VI* – for tenor saxophone and tape
- *Pneuma VII* – for violin and tape
- *Pneuma VIII* – for elec. processed cello and tape
- *Pneuma IX* – for electronics and tape
- *Pneuma X* – for piano and tape
- *Pneuma XI* – for percussion and tape
- *Pneuma XII* – for percussion and tape
- *Phases 1, 2, and 3* – for operatic voice in two parts [to be performed with or without 6 or more Pneumata and/or tape]

Other Works

- *Practical Environments No. 2 (Lower Manhattan)* (2024) – installation work for the Sheen Center, with source material recorded and transmitted below 14th St.
- *El movimiento en la quietud* (2020) – online installation with Ernesto Cárcamo-Cavazos and Esteban Ruiz-Velasco
- *Titus* (2016) – music and sound design for New York Deaf Theater's production of Shakespeare, dir. by Monique Holt
- *Le grand spectacle de l'effort et de l'artifice* (2016) – music for Funsch Dance Experience, co-composed with Ryan Ross Smith
- *Instructions on How to Write Music Nos.1–12* (2016)
- *Augustan Club Waltz* (2015) – arr. 2 marimbas (4 players), and xylo (Scott Joplin [1901])
- *Three Rags for Pianoforte* (2015) – arr. 2 marimbas (4 players), and xylo (James Tenney [1969])
- *The Best Show Ever!* (2013) – instruction-based sculpture
- *ST/v* (2012) – video (dur. 3m)
- *Sonic Trichromacy* (2011) – sound/sculpture installation
- *The Stone Wife* (2009/10) – music/sound design for a play of the same title, dir. Helen Pau
- -*on* (2009) – video (dur. 9m 30s)
- *Film for Radio* (2009) – video (dur. 1m)

Daniel McKemie

Daniel McKemie is a composer, percussionist, programmer, researcher, and arranger based in Oneonta, New York. His work focuses on applying techniques from Music Information Retrieval (MIR) feature extraction and low-level signal processing to explore timbre, integrating that data with original synthesis methods to create new musical works. Additionally, Daniel designs innovative methods for interfacing handmade circuitry, modular synthesizers, and embedded systems with both legacy and contemporary software. This approach enables the creation of complex, responsive performance environments, where software generates compositional processes as analog signals sent to hardware, and analyzes hardware feedback to influence musical behavior in real time.

Daniel currently serves as a Lecturer at SUNY Oneonta, where he teaches courses on electronic music, audio arts, and music history. He earned his MS in Computer Science and Health Informatics at Brooklyn College, where he worked as a graduate research assistant in Professor Johanna Devaney's Laboratory for Understanding Music and Audio (LUMaA). He also served as an adjunct lecturer, teaching courses in computer music and programming.

His music has been performed throughout Europe, Asia, South America, and Australia, and his research on computer music and web-based audio/composition techniques has been presented and published internationally at conferences including the International Computer Music Conference (ICMC), the Korean Electroacoustic Music Society (KEAMS), the Australasian Computer Music Association (ACMA), the International Symposium on Computer Music Multidisciplinary Research (CMMR), the Society for Electroacoustic Music in the United States (SEAMUS), and the Percussive Arts Society International Convention (PASIC), among others.

In addition to his work in electronic music, Daniel is an active percussionist, having premiered dozens of new works, many in collaboration with the William Winant Percussion Group. As an instrumental composer, he has written for soloists, chamber ensembles, mixed media projects, and radio broadcasts. He also served as an assistant transcriber and orchestrator for Roscoe Mitchell on pieces from his *Conversations* collection.

Daniel has studied with Morton Subotnick, Roscoe Mitchell, William Winant, Dean Gronemeier, and Pauline Oliveros, and has collaborated with a wide range of artists and organizations, including the New York Deaf Theatre, Larry Polansky, the Iceland Symphony Orchestra, the Montreal-Toronto Art Orchestra, Funsch Dance Experience, Christian Wolff, Bob Ostertag, and Steve Schick. He also earned his MA in Music Composition from Mills College, and his BA in Music Performance from the University of Nevada, Las Vegas.

This collection brings together fifteen years of notes, essays, and compositions from a working life in experimental music. Originally intended as academic research, it became a personal project driven by curiosity rather than credentials. The texts explore music-making across tools and formats: from live electronics and noise, to systems-based composition, orchestration, arrangement, practice and performance. A record of ideas, questions, and aesthetics. A document for the future.

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Daniel McKemie is a composer, percussionist, programmer, researcher, and arranger based in Brooklyn, New York. His work focuses on applying techniques from Music Information Retrieval (MIR) feature extraction and low-level signal processing to explore timbre, integrating that data with original synthesis methods to create new musical works. Additionally, Daniel designs innovative methods for interfacing handmade circuitry, modular synthesizers, and embedded systems with both legacy and contemporary software.

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