

HYPERCOMPRESSION

Stochastic Musical Processing

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Bachelor of Music

University of Massachusetts Lowell, 2008

Submitted to the Program in Media Arts and Sciences,
School of Architecture and Planning, in partial fulfillment
of the requirements for the degree of

Master of Science in Media Arts and Sciences

at the **Massachusetts Institute of Technology**

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Abstract

The theory of stochastic music proposes that we think of music as a vertical integration of mathematics, the physics of sound, psychoacoustics, and traditional music theory. In *Hypercompression, Stochastic Musical Processing* we explore the design and implementation of three innovative musical projects that build on a deep vertical integration of science and technology in different ways: Stochastic Tempo Modulation, Reflection Visualizer, and Hypercompression. Stochastic Tempo Modulation proposes a mathematical approach for composing previously inaccessible polytempic music. The Reflection Visualizer introduces an interface for quickly sketching abstract architectural and musical ideas. Hypercompression describes new technique for manipulating music in space and time. For each project, we examine how stochastic theory can help us discover and explore new musical possibilities, and we discuss the advantages and shortcomings of this approach.

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1

Introduction

In his 1963 book, *Formalized Music*, Composer, Engineer, and Architect Iannis Xenakis described the foundation for his own reinterpretation of conventional music theory:

“All sound is an integration of grains, of elementary sonic particles, of sonic quanta. Each of these elementary grains has a threefold nature: duration, frequency, and intensity.”

Instead of using high-level musical concepts like pitch and meter to compose music, Xenakis posited that only three elementary qualities (frequency, duration and intensity) are necessary. Because it is impractical to describe sounds as the sum of hundreds or thousands of elementary sonic particles, he proposed using statistical and probabilistic models to define sounds at the macroscopic level, and using similar mathematical models to describe other compositional concepts including rhythm, form, and melody. Xenakis named this new style *stochastic*, and he considered it a generalization of existing music theory: High-level musical constructs such as melody, harmony, and meter are mathematical abstractions of the elementary sonic particles, and alternative abstractions, such as non standard tuning, should exist as equals within the same mathematical framework. Music composition, he claimed, requires a *deep* understanding of the mathematical relationship between sonic elements and musical abstractions, and music composition necessarily involves the formulation of new high-level musical constructs built from the low-level elements.

Music and Mathematics Today Today, just over 50 years after *Formalized Music* was first published, some of Xenakis’ ideals have been widely adopted by musicians and composers. As computers, amplifiers, and electronics become ubiquitous in the composition, production, and performance of music, the line between composer and engineer becomes increasingly indistinct.

As a result, a deep understanding of the mathematics of music is also increasingly valuable to musicians. The most common tools for shaping sounds typically use mathematical language, such as frequency, milliseconds, and decibels, requiring us to translate between mathematical and musical concepts. For example, the interface for a typical electronic synthesizer will include the following controls:

- Attack time, a duration measured in milliseconds
- Filter frequency, measured in cycles per second
- Sustain level, intensity, measured in decibels

Countless high-level interfaces for manipulating sound have been created, but no particular type of abstraction has been widely adopted. It would appear that the most useful (and the most widely used) engineering tools are the simplest:

1. The equalizer, a frequency specific amplifier
2. The delay, a temporal shift on the order of milliseconds
3. The compressor, an automatic gain control

Sound and Space It is curious that the most useful tools for engineering sound would also be the simplest: The compositional equivalent would be composing by starting with pure sine waves as Xenakis originally suggested. From an engineering perspective, Xenakis' three sonic elements are rational choices. By summing together the correct recipe of sine tones, we can construct any audio waveform. However, a waveform is not the same as music, or even the same as sound. Sound is three-dimensional, sound has a direction, and sound exists in space. Could space be the missing sonic element in electronic audio production? If we design our audio engineering tools such that space is considered an equal to frequency and intensity, can we build high-level tools that are as effective as the low level tools we depend on, or even as effective as acoustic instruments? The three projects described in this thesis are directly inspired by Iannis Xenakis, and rest on the foundation of stochastic music: Stochastic Tempo Modulation, Reflection Visualizer, and Hypercompression. Each project builds on existing paradigms in composition or audio engineering, and together they treat space and time as equals and as true elements of music.

1.1 Stochastic Tempo Modulation

Music and time are inseparable. All music flows through time and depends on temporal constructs - the most common being meter and tempo. Accelerating or decelerating tempi are common in many styles of music, as are polyrhythms. Music with multiple simultaneous tempi or *polytempic music* is less common, but still many examples can be found. Fewer examples of music with simultaneous tempi that shift relative to each other exist, however, and it is difficult for musicians to accurately perform changing tempi in parallel. Software is an obvious choice for composing complex and challenging rhythms such as these, but existing compositional software makes this difficult. Stochastic Tempo Modulation offers a solution to this challenge by describing a strategy for composing music with multiple simultaneous tempi that accelerate and decelerate relative to each other. In [chapter 3](#) we derive an equation for smoothly ramping tempi to converge and diverge as musical events within a score, and show how this equation can be used as a stochastic process to compose previously inaccessible sonorities.

1.2 Reflection Visualizer

Music and space are intimately connected. The first project, described in [chapter 4](#), introduces an interface for quickly sketching and visualizing simple architectural and musical ideas. Reflection Visualizer is a software tool that lets us design and experiment with abstract shapes loosely based on two-dimensional acoustic lenses or “sound mirrors.” It is directly inspired by the music and architecture of Xenakis.

1.3 Hypercompression

Time and space are the means and medium of music. Hypercompression explores a new tool built for shaping music in time and space. The tool build on the dynamic range compression paradigm. We usually think of compression in terms of *reduction*: Data compression is used to reduce bit-rates and file sizes, while audio compression is used to reduce dynamic range. Record labels’ use of dynamic range compression as a weapon in the *loudness war*^{1,2} has resulted in some of today’s music recordings utilizing no more dynamic range than a 1909 Edison cylinder.³ A deeper study of dynamic range compression, however, reveals more subtle and artistic applications beyond that of reduction. A

¹ Beginning in the 1990s, record labels have attempted to make their music louder than the music released by competing labels. “Loudness War” is the popular name given to the trend of labels trying to out-do each other at the expense of audio fidelity.

² Emmanuel Deruty and Damien Tardieu. About Dynamic Processing in Mainstream Music. *AES: Journal of the Audio Engineering Society*, 62(1-2):42–56, 2014. ISSN 15494950

³ Bob Katz. *Mastering Audio: The Art and Science*. Focal Press, 2nd edition, 2007. ISBN 978-0240808376

skilled audio engineer can apply compression to improve intelligibility, augment articulation, smooth a performance, shape transients, extract ambience, de-ess vocals, balance multiple signals, or even add distortion.⁴ At its best, the compressor is a tool for temporal shaping, rather than a tool for dynamic reduction.

Hypercompression expands the traditional model of a dynamic range compressor to include spatial shaping. Converting measurement of sound from the cycles per second (in the temporal domain) to wavelength (in the spatial domain) is a common objective in acoustics and audio engineering practices.⁵ While unconventional, spatial processing is a natural fit for the compression model. The mathematics and implementation of the Hypercompressor are described in detail in [chapter 5](#).

⁴ Alex Case. *Sound FX: Unlocking the Creative Potential of Recording Studio Effects*. Focal Press, 2007. ISBN 978-0240520322

⁵ G Davis and R Jones. *The Sound Reinforcement Handbook*. Recording and Audio Technology Series. Hal Leonard, 1989. ISBN 9780881889000

1.4 Performance

Hypercompression was used in the live performance of *De L'Expérience*, a new musical work by composer Tod Machover for narrator, organ, and electronics. During the premier at the Maison Symphonique de Montréal in Canada, Hypercompression was used to blend the electronics with the organ and the acoustic space. A detailed description of how Hypercompression featured in this performance is also discussed in [chapter 6](#).

1.5 Universality

At the MIT Media Lab, we celebrate the study and practice of projects that exist outside of established academic disciplines. The Media Lab (and the media) have described this approach as interdisciplinary, cross-disciplinary, anti-disciplinary, and post-disciplinary; rejecting the cliché that academics must narrowly focus their studies learning *more and more about less and less*, and eventually knowing *everything about nothing*. The projects described here uphold the vision of both Xenakis and the Media Lab. Each chapter documents the motivations and implementation of a new tool for manipulating space and sound. Each project draws from an assortment of fields including music, mathematics, computer science, acoustics, audio engineering and mixing, sound reinforcement, multimedia production, and live performance.

2

Background

In this chapter, we review the precedent for contemporary explorations of time and space in music. While far too many projects exist to cover them all, we focus on projects that are either particularly impactful or particularly relevant to the projects described in this thesis. We conclude with a study of Iannis Xenakis' involvement in the Philips Pavilion at the 1958 Brussels World Fair, which made particularly innovative use of sound and space.

Early Spatial Music Western spatial music emerged during the renaissance period. The earliest published example of spatial music was by Adrian Willaert in 1550.¹ The Basilica San Marco in Venice, where Willaert was *maestro di capella*, had an interesting feature: two separate pipe organs facing each other across the chapel. Willaert took advantage of this unusual setup by composing music for separate choirs and instrumental groups adjacent the two organs. Spatially separate choirs soon became a fashion and gradually spread beyond Venice, as more and more spatially separated groups were incorporated into composition. In honor of Queen Elizabeth's 40th birthday in 1573, Thomas Tallis composed *Spem in alium*, a choral piece with 40 separate parts arranged in eight spatially separated choirs. Interest in spatial composition declined toward the end of the Baroque period, and was largely avoided until the Romantic period. Beriloz' *Requiem* in 1837, Giuseppe Verdi's *Requiem* in 1874, and Mahler's *Symphony No. 2* in 1895 all feature spatially separated brass ensembles.

Tempo Acceleration and Deceleration Chapter 3 is concerned with oblique, similar, and contrary tempo accelerations and decelerations in the context of polytempic (with two or more simultaneous tempi) music. The tempo indicators commonly seen today, such as *allegro* and *adagio*, emerged during the 17th century in Italy. While these markings partly express a mood (*gaily* and *with leisure*,

¹ Richard Zvonar. A History Of Spatial Music, 1999. URL <https://pantherfile.uwm.edu/kdschlei/www/files/a-history-of-spatial-music.html>

Do not to confuse polytempic music with *polymetric* and *polyrhythmic* music. Polyrhythms and polymeters are common in West African musical tradition and appear much earlier in Western music than polytempi.

respectively), rather than a strict tempo, they were much easier to follow than the proportional system (based on tempic ratios such as 3:2 and 5:4) that they replaced.² The intentional use of gradual tempo changes likely evolved from the unconscious but musical tempo fluctuations of a natural human performance. We can see examples of the purposeful manipulation of tempo in the Baroque period. Monteverdi's *Madrigali guerrieri* from 1638 includes adjacent pieces: *Non havea Febo ancora*, and *Lamento della ninfa*. The score instructs to perform the former piece *al tempo della mano* (in the tactus of the conducting hand), and the latter *a tempo del'affetto del animo e non a quello della mano* (in a tempo [dictated by] emotion, not the hand). While Monteverdi's use of controlled tempo was certainly not the first, we are particularly interested in gradual tempo changes in polytempic compositions, which do not appear in Western music until near the beginning of the 20th century.

² C Sachs. *Rhythm and Tempo: A Study in Music History*. W.W. Norton and Company, 1953

2.1 20th Century Modernism

As the Romantic period was coming to an end, there was a blossoming of complexity, diversity, and invention in contemporary music. While Performers developed the virtuosic skills required to play the music, composers also wrote increasingly difficult scores to challenge the performers.³ Works by Italian composer Luciano Berio illustrate the complexity of contemporary music of the time. Beginning in 1958, Berio wrote a series of works he called *Sequenza*. Each was a highly of highly technical composition written for a virtuosic soloist. Each was for a different instrument ranging from flute to guitar to accordion. In *Sequenza IV*, for piano, Berio juxtaposes thirty-second note quintuplets, sextuplets, and septuplets (each with a different dynamic), over just a few measures.

³ D. J. Grout, J. P. Burkholder, and C. V. Palisca. *A History of Western Music*. W.W. Norton, 7th edition, 2006. ISBN 0-393-97991-1

2.2 Polytempic Music

Western polytempi can be traced to Henry Cowell's book, *New Musical Resources*, first published in 1930, wherein Cowell states:

"Rhythm presents many interesting problems, few of which have been clearly formulated. Here, however, only one general idea will be dealt with—namely, that of the relationship of rhythm, which have an exact relationship to sound-vibration, and, through this relationship and the application of overtone ratios, the building of ordered systems of harmony and counterpoint in rhythm, which have an exact relationship to tonal harmony and counterpoint."^{4,5}

⁴ Examples from *New Musical Resources* are from the 3rd edition, published in 1996.

⁵ Henry Cowell. *New Musical Recourses*. Cambridge University Press, Cambridge, 3rd edition, 1996. ISBN 0 421 49651 9

Cowell goes on to describe a system of ratios for tempi: If we think of two parallel tempi, one going twice the rate of the other, it is akin to the octave pitch interval, one fundamental frequency being twice the other. Similarly, the vibration ratio of 2:3 can be thought of as a fifth, and some rhythmic relationships are “harmonious”, while others are “dissonant.” This is nearly identical to the proportional tempo technique that was displaced in Italy in the 1600s, but Cowell does eventually introduce the concept of polytempic music:

“The use of different simultaneous tempi in a duet or quartet in opera, for instance would enable each of the characters to express his individual mood; such a system might effectively be applied to the famous quartet from *Rigoletto*, in which each of the characters is expressing a different emotion.”

This example is closer to what we are interested in, but does not include simultaneous tempo changes. However, Cowell takes the idea a step further, illustrating the possibility of parallel tempo acceleration with figure 2.1.

While he was not a mathematician, Cowell did understand that there were some unanswered complications surrounding simultaneous tempo changes. While describing polytempic accelerations he notes:

“For practical purposes, care would have to be exercised in the use of sliding tempo, in order to control relation between tones in a sliding part with those in another part being played at the same time: a composer would have to know, in other words, what tones in a part with rising tempo would be struck simultaneously with other tones in a part of, say, fixed tempo, and this from considerations of harmony. There would usually be no absolute coincidence, but the tones which would be struck at approximately the same time could be calculated.”

It is possible to calculate exactly when tones in an accelerating tempo will be struck. In the examples shown in figure 2.1, the linear tempo accelerations only rarely yield satisfactory results. Figure 2.1 does not show how many beats or measures elapse during the tempo acceleration, but with linear acceleration shown, the parallel tempi are likely to be out of phase once the tempo transition is complete. This is described in more detail in chapter 3.

Modernism and Rhythmic Complexity

During the Modernist period, many composers sought new ways to use time and space as compositional elements, and polytempic music was relatively unexplored. Traditional music notation is not well-equipped to handle acceleration with precision. The conventional way to describe gradual tempichanges is to annotate the

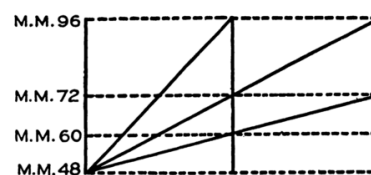


Figure 2.1: Polytempic tempo transitions as illustrated by Henry Cowell in 1930. © Cambridge University Press

score with notes like *ritardando* (gradually slowing) and *accelerando* (gradually accelerating), coupled with traditional Italian tempo markings like *adagio* (slow, stately, at ease) and *allegro* (fast, quickly, bright). Exact tempo rates can be explicitly specified with an M.M.⁶ marking. It is not realistic to expect a performer to be able to follow a precise mathematical acceleration. This did not stop modernist composers from finding creative ways to notate surprisingly precise polytempic compositions using only the conventional notation:

⁶ In a musical score, M.M. stands for Maelzel's Metronome, and is accompanied by a number specifying the beats per minute.

1. Groups of tuplets layered against a global tempo, as used by Henry Cowell (*Quartet Romantic*, 1915-17) and Brian Ferneyhough (*Epicycle for Twenty Solo Strings*, 1968).
2. Polymeters are notated against a global tempo, and the value of a quarter note is the same in both sections, as in Elliott Carter's *Double Concerto for Harpsichord and Piano with Two Chamber Orchestras*, 1961 and George Crumb's *Black Angels*, 1971.
3. Sections are notated without meter. Notes are positioned horizontally on the leger linearly, according to their position in time. Conlon Nancarrow (*Study No. 8 for Player Piano*, 1962) and Luciano Berio (*Tempi Conceriati*, 1958-59).
4. The orchestra is divided into groups, and groups are given musical passages with varying tempi. The conductor cues groups to begin (Pierre Boulez, *Rituel: In Memoriam Maderna*, 1974).
5. One master conductor directs the entrances of auxiliary conductors, who each have their own tempo and direct orchestral sections (Brant Henry, *Antiphony One for Symphony Orchestra Divided into 5 Separated Groups*, 1953).

Charles Ives and The Unanswered Question

One composer, Charles Ives, did write polytempic music before *New Musical Resources* was published. Ives was an American composer whose works were largely overlooked during his lifetime. One of these, his 1908 composition, *The Unanswered Question*, is remarkable in that it incorporates both spatial and polytempic elements. In this piece, the string section is positioned away from the stage, while the trumpet soloist and woodwind ensemble are on the stage. A dialogue between the trumpet, flutes, and strings is written into the music, with the trumpet repeatedly posing a melodic question "*The Perennial Question of Existence*." Each question is answered by the flute section. The first response is synchronized with the trumpet part, but subsequent responses

accelerate and intentionally desynchronize from the soloist. Ives included a note at the beginning of the score which describes the behavior of the "The Answers":

This part need not be played in the exact time position indicated. It is played in somewhat of an impromptu way; if there is no conductor, one of the flute players may direct their playing.

The flutes will end their part approximately near the position indicated in the string score; but in any case, "The Last Question" should not be played by the trumpet until "The Silences" of the strings in the distance have been heard for a measure or two. The strings will continue their last chord for two measures or so after the trumpet stops. If the strings shall have reached their last chord before the trumpet plays "The Last Question", they will hold it through and continue after, as suggested above.

"The Answers" may be played somewhat sooner after each "Question" than indicated in the score, but "The Question" should be played no sooner for that reason.

Ives gave the performers license over the temporal alignment, but he made it clear that the parts should not be played together.

Gruppen

Ives' polytempic compositions from the first half of the 20th century are somewhat of an exception. Polytempi was not widely explored until well after *New Musical Resources* was published. One famous example is Karlheinz Stockhausen's *Gruppen* for three orchestras (1955-57). Managing parallel tempi that come in and out of synchronicity is always a challenge with polytempic music, and Stockhausen found an effective, if heavy-handed, solution with a system of discrete tempo changes. Each of the three orchestras was to have its own conductor, and the conductor would listen for a cue carefully written in one of the other sections. That cue would signal to the conductor to begin beating a silent measure at the new tempo and prepare the new orchestra to begin playing. Stockhausen did not say that he was inspired by *New Musical Resources* directly, but his famous essay *How Time Passes* describes how he chose the tempic ratios used in *Gruppen*. Instead of basing the tempo scales on simple pythagorean relationships, Stockhausen chose the relationships based on the $\sqrt[12]{2}$ ratio of adjacent notes in equal tempered tuning.

Conlon Nancarrow

Conlon Nancarrow is best known for his incredibly complex player piano scores, and is recognized as one of the first composers to realize the potential of technology to perform music beyond

human capacity. Unlike Stockhausen, Nancarrow did acknowledge the influence of Cowell's *New Musical Resources* on his own works. His compositions for the player piano, beginning with *Study for Player Piano No. 21*, did incorporate polytempic accelerations⁷. While some of Nancarrow's compositions do feature many simultaneous tempi, (Study No. 37 features 12 simultaneous tempi)⁸, a rigorous mathematical approach would be required for all 12 tempi to accelerate or decelerate relative to each other and synchronize at pre-determined points. Interestingly, Nancarrow said in a 1977 interview that he was originally interested in electronic music, but the player piano gave him more temporal control.⁹

New Polytempi

The many different approaches to polytempi in modernist music all have one thing in common: They all wrestle with synchronicity. Human performers are not naturally equipped to play simultaneous tempi, and composers must find workarounds that make polytempic performance accessible.

The examples described in this chapter exist in one or more of the following categories:

1. The tempo changes are discrete rather than continuous.
2. The music may suggest multiple tempi, bar lines of parallel measures line up with each other, and the "changing" tempi are within a global tempo.
3. The tempo changes are somewhat flexible, and the exact number of beats that elapse during a transition varies from one performance to another.
4. The tempo acceleration is linear, and parallel parts align only at simple mathematical relationships.

It is not simple to rigorously define parallel tempo curves that accelerate and decelerate continuously relative to each other, and come into synchronicity at strict predetermined musical points for all voices. In [chapter 3](#), we discuss how existing electronic and acoustic music approaches this challenge, and derive a mathematical solution that unlocks a previously inaccessible genre of polytempic music.

⁷ Nancy Yunhwa Rao. Cowell's Sliding Tone and the American Ultramodernist Tradition. *American Music*, 23(3):281–323, 2005

⁸ John Greschak. Polytempo Music, An Annotated Bibliography, 2003. URL <http://www.greschak.com/polytempo/ptbib.htm>

⁹ Charles Amirkhanian. An Interview with Conlon Nancarrow, 1977. URL https://archive.org/details/AM_1977_04_28

2.3 Amplified Spatial Music

The evolution of polytempic music in the Modernist period was paralleled by innovation in the creation and performance of electronic music. After World War II, new technology became available to composers, and with this new technology came new styles of music. Pierre Schaeffer was among the first composers using electronic sounds together with acoustic ones. He worked at the Radiodiffusion-Télévision Française (RTF), where he helped to pioneer early practices in *musique concrète*. With the help of younger composer Pierre Henry, he was also among the first composing spatialized pre-recorded sound. The pair collaborated on a piece called *Symphonie pour un Homme Seul* (Symphony for One Man Alone, 1950). For this piece they created a tetrahedral loudspeaker arrangement and a rather dramatic interface that they called the *Pupitre d'espace*, shown in figure 2.2. Large hoops on the device had inductive coils that sensed the user's hand position and controlled the signal routing to the loudspeakers.¹⁰

Gesang der Jünglinge

The success of Schaeffer's music at the RTF attracted the attention of other composers interested in electronic music. Among them was Karlheinz Stockhausen. Stockhausen came to RTF and composed just one 2-track etude in 1952 before returning to West Germany, where he continued to compose orchestral and electronic works. His practice led to the composition of what is widely regarded as the first masterpiece of electronic music, *Gesang der Jünglinge* (Song of the Youths, 1955-56), which was also the first multichannel pre-recorded composition to be performed in a concert setting with multiple loudspeakers placed around the audience¹¹.

The piece stands out by the many aspects in which it is both evolutionary and revolutionary when juxtaposed with the other electronic compositions of the time; the delicate blending of the voice with electronics, and the creative editing of the voice being two examples. It has been extensively analyzed and reviewed in literature;¹² however, the exact strategy for the spatialization of sound in the original four-channel performance remains somewhat ambiguous. From interviews and essays with Stockhausen, we can gather some insight into his process. In 1955, the year when Stockhausen began work on *Gesang der Jünglinge*, he published an essay on his serial technique.

"By regulating the positions of the sources of sound it will be possible for the first time to appreciate aesthetically the universal

¹⁰ Thom Holm. *Electronic and Experimental Music: Technology, Music, and Culture*. Routledge, 2008. ISBN 0415957826



Figure 2.2: Pierre Schaeffer with the *Pupitre d'espace* in 1951. © Ina/Maurice Lecardent, Ina GRM Archives

¹¹ D. J. Grout, J. P. Burkholder, and C. V. Palisca. *A History of Western Music*. W.W. Norton, 7th edition, 2006. ISBN 0-393-97991-1

¹² Pascal Decroupet, Elena Ungeheuer, and Jerome [Translator] Kohl. Through the sensory looking-glass: The aesthetic and serial foundations of *Gesang der Jünglinge*. *Perspectives of New Music*, 36(1):97-142, 1998. ISSN 0031-6016; David Joel Metzger. The Paths from and to Abstraction in Stockhausen's *Gesang der Jünglinge*. *Modernism/modernity*, 11(4):695-721, 2004. ISSN 1080-6601. DOI: 10.1353/mod.2005.0012; and Paul Miller. *Stockhausen and the Serial Shaping of Space*. PhD thesis, University of Rochester, 2009

realisation of our integral serial technique."¹³

In an interview published in 1974 he made the following comment on the subject of sound positioning in *Gesang der Jünglinge*:

"The speed of the sound, by which one sound jumps from one speaker to another, now became as important as pitch once was. And I began to think in intervals of space, just as I think in intervals of pitch or durations. I think in chords of space."¹⁴

A side effect of serialism is discouraging the uneven distribution of a musical parameter. With this in mind, spatialization is a very natural target for serialism. Given the added creative flexibility of surround sound, it reasonable to search for ways to take full advantage of the new dimension, without favoring any particular direction or loudspeaker.

¹³ Karlheinz Stockhausen. *Actualia. Die Reihe*, pages 45–51, 1957

¹⁴ Karlheinz Stockhausen and Jonathan Cott. *Stockhausen : conversations with the composer*. Pan Books, 1974. ISBN 0330241656

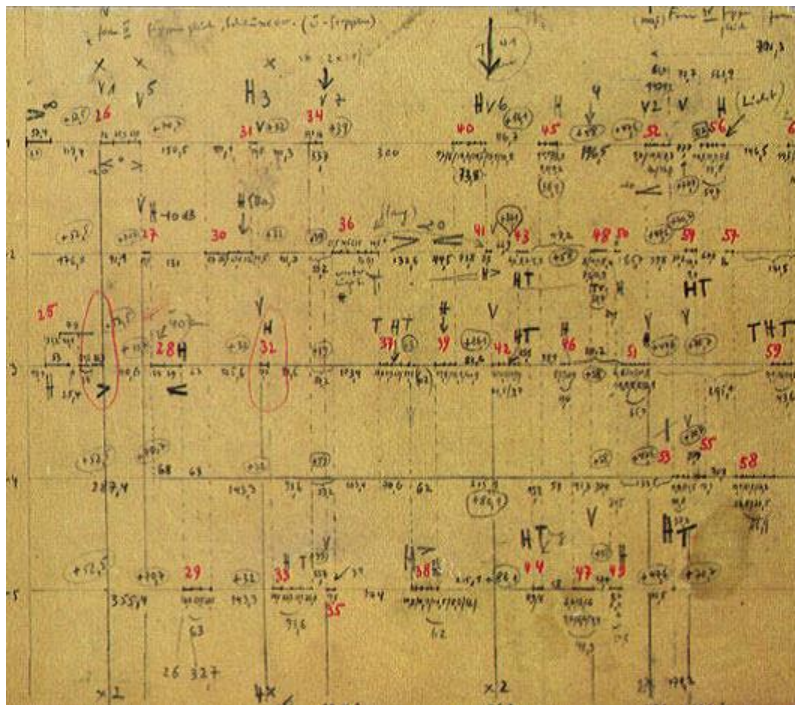


Figure 2.3: Excerpt from the *Gesang der Jünglinge* manuscript.
© www.karlheinzstockhausen.org

Advances in Surround Panning

For his next four-track tape composition, *Kontakte* (1958–60), Stockhausen devised a new technology that made it quite simple to continuously pan sounds between the speakers orbiting the listener. He used a rotating speaker on a turntable, surrounded by four equally spaced microphones. Stockhausen continued to feature spatialization prominently in both acoustic and electronic

work. His major orchestral compositions, *Gruppen* (1955-57, described in section 2.2) and *Carré* (for four orchestra and four choirs, 1959-60) both prominently feature spatialization.

Throughout the rest of the century, advances in technology enabled new performances with more speakers and more complex spatial possibilities. The Vortex multimedia program at the Morrison Planetarium in San Francisco (1957-59) featured 40 loudspeakers with surround sound panning, facilitated by a custom rotary console, and featured works by Stockhausen, Vladimir Ussachevsky, Toru Takemitsu, and Luciano Berio. The planetarium featured synchronized lighting, which became a hallmark of major surround sound productions of the time. The Philips Pavilion at the 1958 Brussels Worlds' Fair used a custom sequencer hooked up to a telephone switcher to pan sounds between over 300 speakers (more in section 2.4). John Chowning's *Turenas*, (1972) simulated amplitude changes and doppler shift of sound objects' movements as a compositional element.¹⁵ The West German pavilion at Expo 70 in Osaka, Japan, included a dome 28 meters in diameter, five hours of music composed by Stockhausen, and 20 soloist musicians. Stockhausen "performed" the live three-dimensional spatialization from a custom console near the center of the dome (figure 2.4). The West German dome was not the only massive spatialized sound installation at Expo 70. Iannis Xenakis, the mastermind behind the 1958 Philips Pavilion in Brussels, was also presenting his 12-channel tape composition, *Hibiki Hana Ma*, at the Japanese Steel Pavilion through 800 speakers positioned around the audience, overhead, and underneath the seats.

¹⁵ John Chowning. *Turenas : the realization of a dream* 1 Introduction 2 Moving Sound Sources and Score. In *Journées d'Informatique Musicale Université de Saint-Etienne*, 2011

Evolution of Electronic Composition

Gesang der Jünglinge may have been the first masterpiece of electronic music, but the techniques that were developed at the RTF studios were quickly spreading. Another composer who was drawn to RTF (where musique concrète was first conceived by Pierre Schaeffer) was Pierre Boulez. However, Boulez was generally unsatisfied with his early electronic compositions and frustrated by the equipment required to make electronic music. Despite his general distaste for electronic music composition, Boulez was approached by the French President, Georges Pompidou, in 1970 and asked to found an institution dedicated to the research of modern musical practice. The center, IRCAM, opened in 1977 with Boulez at the head. In a 1993 interview, Boulez described how he directed the efforts of the lab:

"Back in the 1950s, when you were recording sounds on tape and



Figure 2.4: Inside the West German pavilion at Expo 70. Osaka, 1970.

using them in a concert, you were merely following the tape, which became very detrimental to the performance. So I pushed the research at IRCAM to examine the use of live electronics, where the computer is created for the concert situation, instantly responding to your actions. The system's language also became easier to follow; I remember when I tried to learn the electronics, it was all figures, figures, figures. These meant nothing at all to the musician. If you have to work in hertz and not notes, and then wait half an hour to process the sounds, you get completely discouraged. My goal was so that the musician could sketch his ideas very rapidly, with instantaneous sound and graphical notation. The use of computers finally brought electronics down to the level of understanding for composers. I feel very responsible for that change."¹⁶

Boulez' first major composition that took advantage of the resources at IRCAM was *Répons* which premiered at the Donaueschingen Festival in Germany in 1981 (although Boulez continued to revise it until 1984). The piece balances 24 acoustic performers with pre-recorded material and live processing with spatialization over a ring of 38 loudspeakers. The audience sits in a circle surrounding the orchestra, while six of the acoustic instrumentalists are spaced around the outside of the audience. Boulez was certainly not the first composer to mix electronics with acoustic performers (Milton Babbitt's 1964 *Philomel* is a much earlier example), but *Répons* does mark a certain maturity of the form.

2.4 Iannis Xenakis

The projects in this thesis build on the work and ideas of Iannis Xenakis. Xenakis studied music and engineering at the Polytechnic Institute in Athens, Greece. By 1948, he had graduated from the university and moved to France where he began working for the French architect, Le Corbusier. The job put his engineering skills to use, but Xenakis also wanted to continue studying and writing music. While searching for a music mentor, he approached Oliver Messiaen and asked for advice on whether he should study harmony or counterpoint. Messiaen was a prolific French composer known for rhythmic complexity. He was also regarded as a fantastic music teacher, and his students included Stockhausen and Boulez. Messiaen later described his conversation with Xenakis:

"I think one should study harmony and counterpoint. But this was a man so much out of the ordinary that I said: No, you are almost 30, you have the good fortune of being Greek, of being an architect and having studied special mathematics. Take advantage of these things. Do them in your music."¹⁷

In essence, Messiaen was rejecting Xenakis as a student, but we

¹⁶ Andy Carvin. The Man Who Would be King: An Interview with Pierre Boulez, 1993. URL <http://edwebproject.org/boulez.html>

¹⁷ Tom Service. A Guide to Iannis Xenakis's Music, 2013. URL <http://www.theguardian.com/music/tomserviceblog/2013/apr/23/modern-music-guide-xenakis>

can see how Xenakis ultimately drew from his disparate skills in his compositions. The score for his 1945 composition *Metastasis* (figure 2.5) resembles an architectural blueprint as much as it does a musical score.

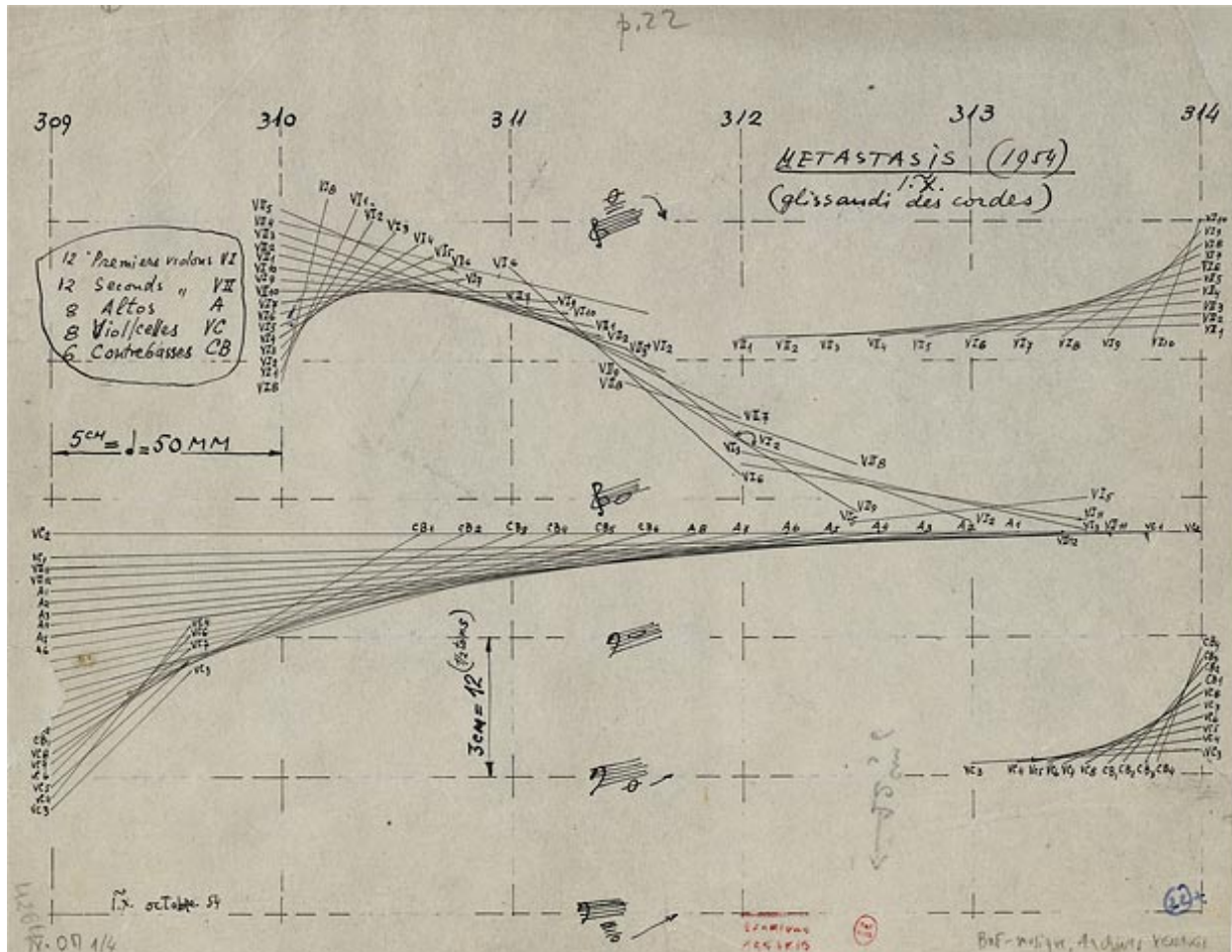


Figure 2.5: Excerpt from Iannis Xenakis' composition, *Metastasis* (1954), measures 309-314. This score in this image was then transcribed to sheet music for the orchestral performance.

The Philips Pavilion

In 1956, Le Corbusier was approached by Louis Kalff (Artistic Director for the Philips corporation) and asked to build a pavilion for the 1958 World's Fair in Brussels. The pavilion was to showcase the sound and lighting potential of Philips' technologies. Le Corbusier immediately accepted, saying:

"I will not make a pavilion for you but an Electronic Poem and a vessel containing the poem; light, color image, rhythm and sound joined together in an organic synthesis."¹⁸

¹⁸ Oscar Lopez. AD Classics: Expo '58 + Philips Pavilion / Le Corbusier and Iannis Xenakis, 2011. URL <http://www.archdaily.com/157658/ad-classics-expo-58-philips-pavilion-le-corbusier-and-iannis-xenakis/>



Figure 2.6: The Philips Pavilion at the 1958 Brussels World Fair as shown in Volume 20 of the *Philips Technical Review*, 1959.

The final product lived up to Le Corbusier's initial description. It included:¹⁹

1. A concrete pavilion, designed by architect and composer Iannis Xenakis
2. *Interlude Sonoire* (later renamed *Concret PH*), a tape music composition by Iannis Xenakis, approximately 2 minutes long, played between performances, while one audience left the pavilion and the next audience arrived
3. *Poème Électronique*, a three-channel, 8 minute tape music composition by composer Edgar Varèse
4. A system for spatialized audio across more than 350 loudspeakers distributed throughout the pavilion
5. An assortment of colored lighting effects, designed by Le Corbusier in collaboration with Philips' art director, Louis Kalff
6. Video consisting mostly of black and white still images, projected on two walls inside the pavilion
7. A system for synchronizing playback of audio and video, with light effects and audio spatialization throughout the experience

Role of Iannis Xenakis During the initial design stage, Le Corbusier decided that the shape of the pavilion should resemble a stomach, with the audience entering through one entrance and exiting out another. He completed initial sketches of the pavilion layout and then delegated the remainder of the design to Xenakis.²⁰

¹⁹ Vincenzo Lombardo, Andrea Valle, John Fitch, Kees Tazelaar, and Stefan Weinzierl. A Virtual-Reality Reconstruction of Poème Based on Electronique Philological Research. *Computer Music Journal*, 33(2):24–47, 2009

²⁰ Joseph Clarke. Iannis Xenakis and the Philips Pavilion. *The Journal of Architecture*, 17(2):213–229, 2012. ISSN 1360-2365. DOI: 10.1080/13602365.2012.678641

The architectural evolution of the pavilion from Le Corbusier's early designs (figure 6.1) to Xenakis' iterations (figure 2.9), illustrates the profound impact that Xenakis had on the project. Xenakis was aware that parallel walls and concave spherical walls could both negatively impact audio perceptibility due to repeated or localized acoustic reflections. The walls of the pavilion had to accommodate lighting effects, which were projected from many different angles, leading him to consider surfaces with a varying rate of curvature.²¹

Ruled surfaces such as the conoid and hyperbolic paraboloid, seemed to meet the needs of the project, and also accommodate the acoustical needs. Through this process, we see Xenakis utilizing the skills that he learned at the Polytechnic Institute and continued to develop while working with Le Corbusier. He also understood the mathematical formation of the ruled surfaces that make up the structure. These surfaces even look familiar to the *Metastasis* score (figure 2.5). In his 1963 book, *Formalized Music*, Xenakis explicitly states that the Philips Pavilion was inspired by his work on *Metastasis*.

2.5 Architecture and Music in Space and Time

In *Formalized Music*²², Xenakis describes how developments in music theory mimic equivalent developments in philosophy, mathematics, and the sciences. Plato, for example, believed that all events transpire as determined by cause and effect. While Plato and Aristotle both described causality in their writing, it was not until the 17th century that controlled experiments and mathematics corroborated the theory.²³ Similarly, music theory has historically employed causal rules to describe counterpoint, tonality, and harmonic movement (such as the natural progression of dominant chord to the tonic).

Causality was largely used to describe physical phenomena until the 19th century when statistical theories in physics began to include probabilistic notions.²⁴ Xenakis noticed that more contemporary fields like *probability theory* generalize and expand on the antecedent theories of causality. Xenakis thought that music composition should naturally follow the progression that physics did, with music theory generalizing and expanding on causal rules that had existed previously. Indeed, starting in the late 19th century and early 20th century, composers like Strauss and Debussy began to bend the existing rules of music theory, composing music that branched away from the causal and tonal theories of the time. With the rise of serialism²⁵ and indeterminate music²⁶, composers such as Stockhausen, Boulez, John Cage,

²¹ S. Gradstein, editor. *The Philips Pavilion at the 1958 Brussels World Fair*, volume 20. Philips Technical Review, 1959

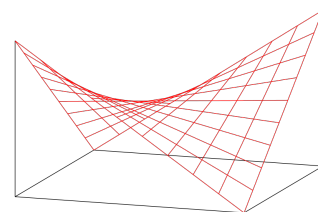


Figure 2.7: A ruled surface. For a surface to be considered "ruled" every point on the surface must be on a straight line, and that line must lie on the surface. In Xenakis' time, ruled surfaces were useful in architecture, because they simplified the construction of curved surfaces by using straight beams.

²² Iannis Xenakis. *Formalized Music*. Pendragon Press, 1992. ISBN 0-945193-24-6

²³ In 1687, Isaac Newton published *Philosophiæ Naturalis Principia Mathematica* (*Mathematical Principles of Natural Philosophy*), in which he compiled the 3 laws of motion that set the foundation for the study of *classical mechanics*.

²⁴ The Maxwell-Boltzmann distribution, which was first derived by James Clerk Maxwell in 1860, describes the probability distribution for the speed of a particle within an idealized gas. For more see <http://plato.stanford.edu/entries/statphys-statmech/>

²⁵ Serialism is a technique for musical composition in which instances of musical elements (such as pitch, dynamics, or rhythm), are given numerical values. Sequences built from the values are ordered, repeated and manipulated throughout the composition.

²⁶ In music, indeterminacy refers to the use of chance (such as rolling dice or flipping coins) as part of the compositional process.

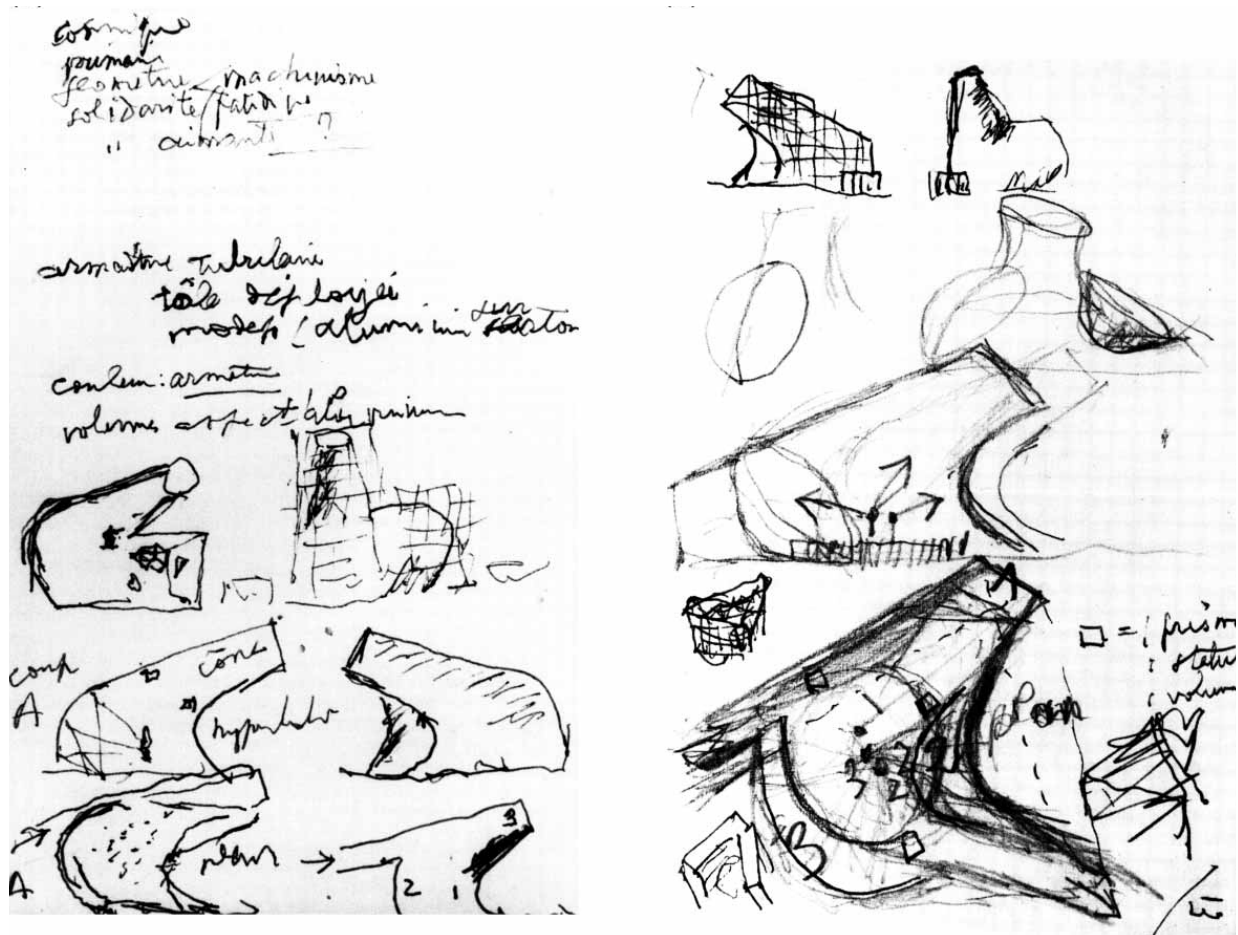


Figure 2.8: Le Corbusier's design sketches for the Philips Pavilion, September – October, 1956 (© 2012 Artists Rights Society, New York/ADAGP, Paris/FLC)

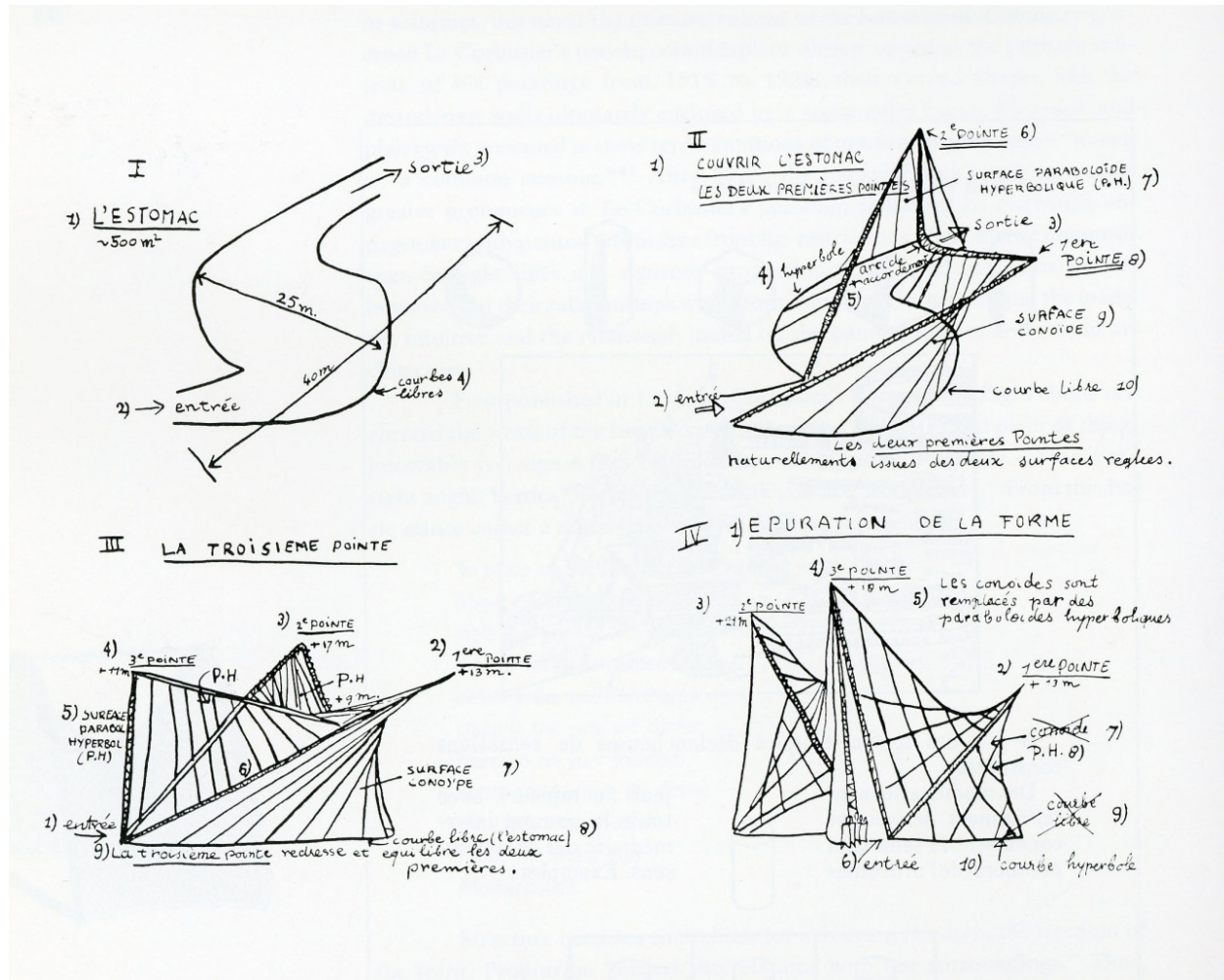


Figure 2.9: Xenakis' early drawings of the Philips Pavilion as documented in volume 20 of the *Philips Technical Review*.

Aaron Copland, and Béla Bartók began to use probability and chance in composition, the same way that physicists were using probability to describe the material world.

To Xenakis' mind, serial music was no less causal than the music it intended to supersede. He described serial music as embodying "virtually absolute determinism."²⁷ Xenakis saw music theory as a sub-set of mathematics and algebra: While musicians have a different vocabulary, they also use mathematical principles to describe and compose music. Because Xenakis understood mathematics as well as music, he was able to identify how even in serialism and indeterminate music, composers were only utilizing a small subset of algebraic theory. In his own music, Xenakis wanted to generalize and expand the causal framework that musicians and theorists had been using to compose and

²⁷ Iannis Xenakis. *Formalized Music*. Pendragon Press, 1992. ISBN 0-945193-24-6

understand music, paralleling similar developments in physics and mathematics. As a reference to *chance*, or *stochos*, Xenakis coined the term *stochastic music* to describe his development.

Xenakis' book, *Formalized Music* gives a verbose explanation of stochastic music. Some authors have interpreted his description more explicitly. In *Audible Design*, Trevor Wishart describes the stochastic process used to compose stochastic music as:

"A process in which the probabilities of proceeding from one state, or set of states, to another, is defined. The temporal evolution of the process is therefore governed by a kind of weighted randomness, which can be chosen to give anything from an entirely determined outcome, to an entirely unpredictable one."²⁸

²⁸ Trevor Wishart. *Audible Design*. Orpheus The Pantomime Ltd., 1994. ISBN 0951031317, 9780951031315

Xenakis' Reflection In the Spring of 1976, while defending his doctoral thesis at the University of Paris, Xenakis emphasized the relevance of seemingly unrelated disciplines to the creative process. A translation of his defense includes this statement:

"The artist-conceptor will have to be knowledgeable and inventive in such varied domains as mathematics, logic, physics, chemistry, biology, genetics, paleontology (for the evolution of forms), the human sciences, and history; in short, a sort of *universality*, but one based upon, guided by and oriented toward forms and architectures."²⁹

²⁹ L Russolo. *The Art of Noises*. Monographs in musicology. Pendragon Press, 1986. ISBN 9780918728227

From Xenakis' drawings we can deduce that he used the same tools, skills, and philosophy to imagine and conceive both music and architecture. His approach elevated both forms and blurred the distinction between the two. Perhaps if we had kept using pen and paper to design buildings and write music, the reality today would be closer to the ideal that he imagined.

As the ideas that inspired Xenakis and other progressive 20th century composers were taking root in contemporary music, the culture of artistic form and composition was already beginning the transition into the digital domain. There is no reason why digital tools cannot favor stochastic processes to linearity, but software for composing music tends to favor static pitches to glissandi, while software for architectural design tends to favor corners to curves. This is where the projects described here make a contribution. By drawing from music, mathematics, computer science, acoustics, audio engineering and mixing, sound reinforcement, multimedia production, and live performance, we can create tools that allow us to indiscriminately compose with space and sound.

3

Temporal Domain: Stochastic Tempo Modulation

One composer writing rhythmically complex music during the 20th century was Elliott Carter. Carter developed a technique he called *tempo modulation*, or *metric modulation* in which his music would transition from one musical meter to another through a transitional section that shared aspects of both. While metric modulation is a technique for changing meter, and Stochastic Tempo Modulation is a technique for changing tempo, the former led to the later in a surprising way. Carter's reputation for complexity in music attracted the attention of composer and cellist Tod Machover. While Machover was studying with Carter, he wrote a trio for violin, viola, and cello, in which each instrument would accelerate or decelerate relative to the others. The piece turned out to be so difficult that it was impossible to find anyone who could play it correctly. Faced with this challenge, Machover saw opportunity:

"A sort of lightbulb went off... computers are out there, and if you have an idea and can learn how to program, you should be able to model it."¹

If the music is too complex for a human to process, but we can define it formulaically, we can teach a computer to play sounds that a human cannot. Stochastic Tempo Modulation builds on this idea with inspiration from Xenakis.

3.1 *Stochos*

In [chapter 1](#) (see [figure 2.5](#)) we saw how Xenakis used ruled surfaces in his composition to compose swarms of notes that move together, creating stochastic sonorities. The goal of Stochastic Tempo Modulation is to enable composition with swarms of tempo modulations that move in correlated, cohesive patterns. Music with two or more simultaneous tempos (polytempic music) is itself not a new concept; many examples exist,² and were

¹ Evan Fein. Q&A With Tod Machover, 2014. URL <http://www.juilliard.edu/journal/1402/hyperinstruments-crowd-sourced-symphonies>

² John Greschak. Polytempo Music, An Annotated Bibliography, 2003. URL <http://www.greschak.com/polytempo/ptbib.htm>

described in [chapter 2](#). Less common is polytempic music where continuous tempo accelerations or decelerations are defined relative to each other. This style of music is well-suited to tape music, because tape machines can play recordings back at variable rates. However, it is difficult to control the exact point (or phase) when de-synchronized tape becomes re-aligned. Performative music with simultaneous tempi that accelerate and decelerate relative to each other is unusual, but does exist. In a 1971 interview composer Steve Reich described how he made the transition to performative polytempic music after working on his tape music composition, *Come Out*:

“1966 was a very depressing year. I began to feel like a mad scientist trapped in a lab: I had discovered the phasing process of *Come Out* and didn’t want to turn my back on it, yet I didn’t know how to do it live, and I was aching to do some instrumental music. The way out of the impasse came by just running a tape loop of a piano figure and playing the piano against it to see if in fact I could do it. I found that I could, not with the perfection of the tape recorder, but the imperfections seemed to me to be interesting and I sensed that they might be interesting to listen to.”³

³ Michael Nyman. Steve Reich. *The Musical Times*, 112:229–231, 1971

Reich’s experience illustrates what other composers and performers have also encountered: It is quite difficult to perform polytempic music accurately. In *Piano Phase*, Reich has two performers playing the same 12 tone series on the piano. After a set number of repetitions through the pattern, one performer begins to play slightly faster until she is exactly one note ahead of the other performer, at which point both performers play at the same rate for a time. This process is repeated and iterated on, creating a live *phasing* effect without the pitch shifting that would occur when phasing analog tape. If we compare a live performance⁴ with a programmatic rendering⁵ of *Piano Phase*, we can hear how the latter is able to accelerate more smoothly. The programmatic example spends longer on the transitions where the two parts are out of phase.

⁴ Tine Allegaert and Lukas Huisman. Reich: Piano Phase, 1989. URL www.youtube.com/watch?v=i0345c6zNfM

⁵ Alexander Chen. Pianophase.com, 2014. URL <https://vimeo.com/98823512>

3.2 Objective

Steve Reich composed *Piano Phase* for two performers. Through experimentation, he found that if the music is reasonably simple, two performers can make synchronized tempo adjustments relative to each other well enough to yield compelling results. Stochastic Tempo Modulation allows us to write music with many more simultaneous tempi. However, the requirements are probably too demanding for unassisted performers. Our goal is to compose and audition music where:

1. Swarms of an arbitrary number of simultaneous tempi coexist.
2. Each individual player within the swarm can continuously accelerate or decelerate individually, but also as a member of a cohesive whole.
3. Each musical line can converge and diverge at explicit points.
At each point of convergence the phase of the meter within the tempo can be set.

We start by defining a single tempo transition. Consider the following example (shown in figure 3.1):

- Assume we have 2 snare drum players. Both begin playing the same beat at 90 BPM in common time.
- One performer gradually accelerates relative to the other. We want to define a continuous tempo curve such that one drummer accelerates to 120 BPM.
- So far, we can easily accomplish this with a simple linear tempo acceleration. However, we want the tempo transition to complete exactly when *both* drummers are on a down-beat, so the combined effect is a 3 over 4 rhythmic pattern.
Linear acceleration results in the transition completing at an arbitrary phase.
- We want the accelerating drummer to reach the new tempo after exactly 20 beats.
- We also want the acceleration to complete in exactly 16 beats of the original tempo, so that the drummer playing a constant tempo and the accelerating drummer are playing together.



Figure 3.1: Tempo transition from 90 BPM to 120 BPM

3.3 Solution

We are interested in both the number of beats elapsed in the static tempo *and* in the changing tempo, as well as the absolute tempo.

If we think of the number of beats elapsed as our *position*, and the tempo as our *rate*, we see how this resembles a physics problem. If we have a function that describes our tempo (or rate), we can integrate that function, and the result will tell us our number of beats elapsed (or position). Given the above considerations, our tempo curve is defined in terms of 5 constants:

- Time $t_0 = 0$, when the tempo transition begins
- A known time, t_1 , when the tempo transition ends
- A known starting tempo: \dot{x}_0
- A known finishing tempo: \dot{x}_1
- The number of beats elapsed in the changing tempo between t_0 and t_1 : x_1

The tension of the tempo curve determines how many beats elapse during the transition period. The curve is well-defined for some starting acceleration a_0 and finishing acceleration a_1 , so we define the curve in terms of linear acceleration. Using Newtonian notation we can describe our tempo acceleration as:

$$\ddot{x}_1 = a_0 + a_1 t_1 \quad (3.1)$$

Integrating linear acceleration (3.1) yields a quadratic velocity curve (3.2). The velocity curve describes the tempo (in beats per minute) with respect to time.

$$\dot{x}_1 = \dot{x}_0 + a_0 t_1 + \frac{a_1 t_1^2}{2} \quad (3.2)$$

We must specify the same time units for input variables like t_1 and \dot{x}_1 . I prefer *minutes* for t_1 and *beats per minute* for \dot{x}_1 over *seconds* and *beats per second*.

Integrating velocity (3.2) gives us a function describing position (the number of beats elapsed with respect to time).

$$x_1 = x_0 + \dot{x}_0 t_1 + \frac{a_0 t_1^2}{2} + \frac{a_1 t_1^3}{6} \quad (3.3)$$

With equations (3.2) and (3.3), we can solve for our two unknowns, a_0 and a_1 . First we solve both equations for a_1 :

$$a_1 = \frac{-2}{t_1^2}(\dot{x}_0 - \dot{x}_1 + a_0 t_1) = \frac{-6}{t_1^3}(\dot{x}_0 - \dot{x}_1 + \frac{a_0 t_1^2}{2})$$

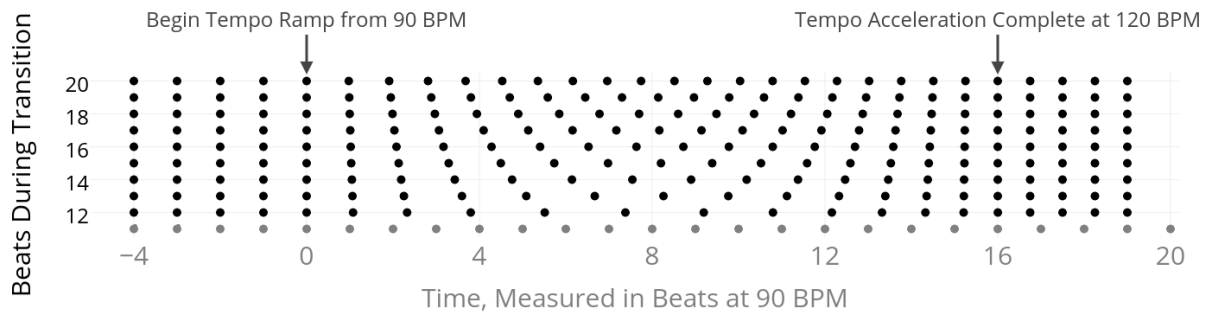
Assuming $t_1 \neq 0$, we solve this system of equations for a_0 :

$$a_0 = \frac{6\dot{x}_1 - 2t_1(\dot{x}_1 + 2\dot{x}_0)}{t_1^2} \quad (3.4)$$

Evaluating (3.4) with our constants gives us our starting acceleration. Once we have a_0 we can solve (3.2) for a_1 , and evaluate (3.2) with a_1 and a_0 to describe our changing tempo with respect to time.

3.4 Stochastic Transitions

Equipped with the equations from the previous section, it becomes quite simple to create swarms of parallel tempos that are correlated and complex. In figure 3.2, we build on the previous example. Here, each additional tempo curve is calculated the same way, except for x_1 (number of beats in our accelerating tempo during the transition), which is incremented for each additional tempo line.



This pattern clearly exhibits controlled chance that Xenakis would describe as *stochastic*. On the very first beat at $t = 0$, all parallel parts are aligned. Beats 2 and 3 can be heard as discrete rhythmic events, but become increasingly indistinct. The end of beat 4 then overlaps with the start of beat 5, before articulated beats transition to pseudo random noise. By beat 13 of the static tempo, the chaos of the many accelerating tempi begin to settle back into order before returning to complete synchronicity at $t = 16$.

Figure 3.2: Stochastic Tempo Transition from 90 BPM to 120 BPM. Black dots are beats in our changing tempi. Grey dots show a continuation of beats at the initial tempo. $12 \leq x_1 \leq 20$

3.5 Tide: Composition with Stochastic Tempo Modulation

An earlier version of the equation derived here was packaged as a patch for the Max⁶ graphical programming language. Composer Bryn Bliska developed a user interface and used it in the composition of *Tide*.⁷ While this earlier version of the equation did not support a variable x_1 parameter, *Tide* uses Stochastic Tempo Modulation to drive the phasing tempos of the bell-like synthesizers throughout the piece for two to three simultaneous tempi.

⁶ <https://cycling74.com/products/max/>

⁷ Available online: http://web.media.mit.edu/~holbrow/mas/Tide_Bliska_Holbrow.wav

3.6 Recent Polytempic Work

Many commercial and research projects deal with different ways to manipulate rhythm and tempo. Flexible digital audio worksta-

tions (DAWs) like Cockos Reaper⁸ and MOTU Digital Performer⁹ include features for auditioning tracks or music-objects with unique simultaneous tempi, and individual tempos can even be automated relative to each other. However, the precise non-linear tempo curves that are required for the syncopated musical content to synchronize correctly after a transition completes are not possible in any DAW we tried. Audio programming languages like Max and SuperCollider¹⁰ could be used to create tempo swarms, but require equations like the ones defined in section 3.3. One project, *Realtime Representation and Gestural Control of Musical Polytempi*¹¹ demonstrates an interface for generating Polytempic music, but is not intended or capable of generating coordinated or stochastic tempi swarms. *The Beatbug Network*¹² is described as a multi-user interface for creating stochastic music, but is focused on “beats,” or musical rhythmic patterns, and timbres, rather than tempi. *Stochos*¹³ is a software synthesizer for generating sound using random mathematical distributions, but is also not designed to work with simultaneous tempos or even as a rhythm generator. Finally, *Polytempo Network*¹⁴ is a project that facilitates the performance of polytempic music, but does not aid the composition thereof.

⁸ <http://www.reaper.fm>

⁹ <http://www.motu.com/products/software/dp>

¹⁰ <http://supercollider.github.io/>

¹¹ Chris Nash and Alan Blac. Realtime Representation and Gestural Control of Musical Polytempi. In *New Interfaces for Musical Expression*, pages 28–33, Genova, Italy, 2008

¹² Gil Weinberg, Roberto Aimi, and Kevin Jennings. The Beatbug Network - A Rhythmic System for Interdependent Group Collaboration. In *Proceedings of the International Conference on New Interfaces for Musical Expression*, pages 186–191, Dublin, Ireland, 2002. URL http://www.nime.org/proceedings/2002/nime2002_186.pdf

¹³ Sinan Bokesoy and Gerard Pape. Stochos: Software for Real-Time Synthesis of Stochastic Music. *Computer Music Journal*, 27(3):33–43, 2003. ISSN 0148-9267. DOI: 10.1162/014892603322482510

¹⁴ Philippe Kocher. Polytempo Network: A System for Technology-Assisted Conducting. *Proceedings of the International Computer Music Conference*, 163 (September):532–535, 2014

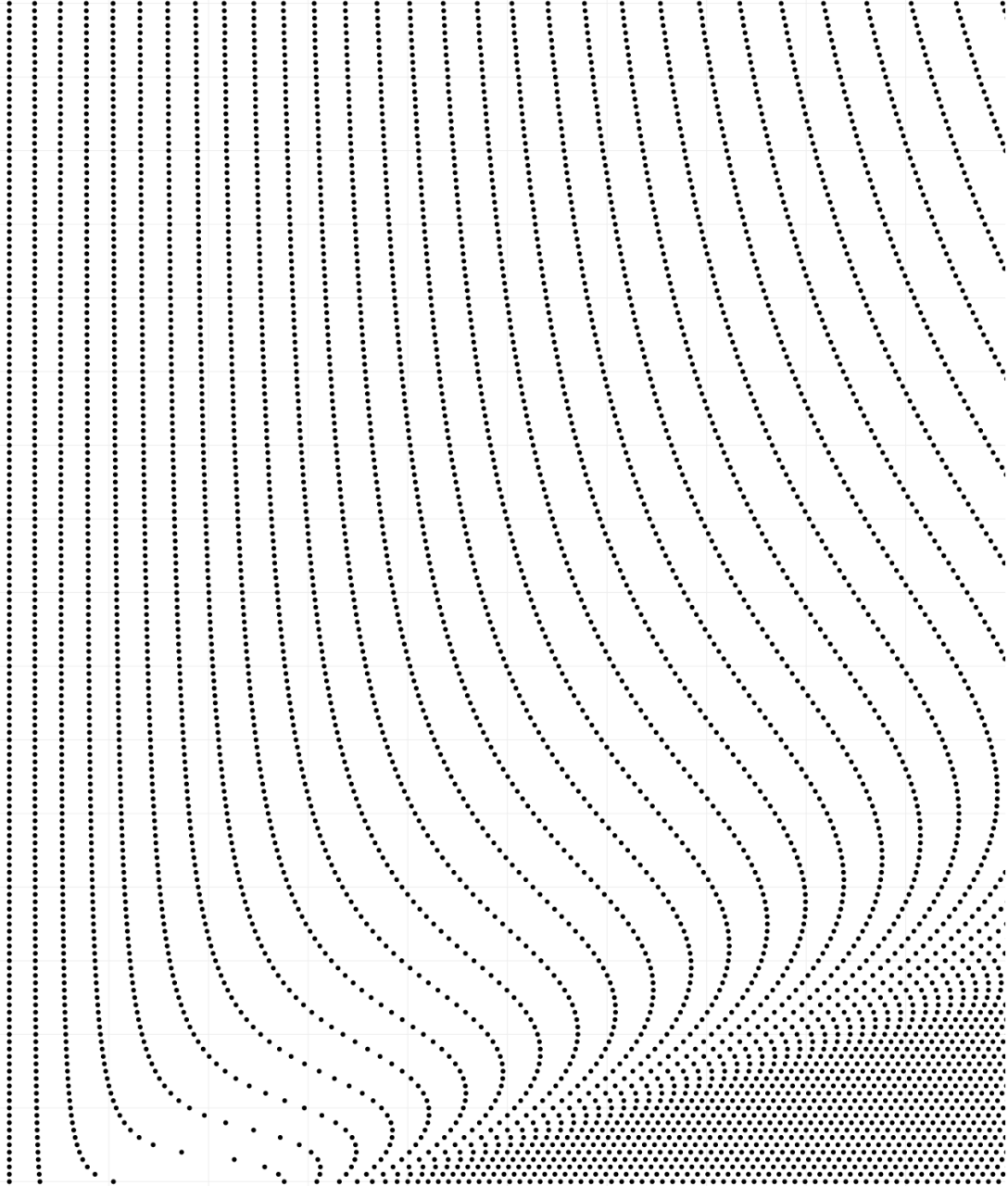


Figure 3.3: Stochastic Tempo Modulation with variable t_1 , variable x_1 , and 161 simultaneous tempi.

4

Spatial Domain: Reflection Visualizer

It was Xenakis' goal for the curved surfaces of the Philips Pavilion to reduce the sonic contribution of sound reflections as much as possible.¹ He knew that reflections and the resulting comb filtering could impair intelligibility and localization of music and sounds. The pavilion was to have hundreds of loudspeakers, and large concave surfaces like the ones on the inside of the pavilion can have a focussing effect on acoustic reflections, resulting in severe filtering and phase cancellations.² If Xenakis had been able to model the reflections and compose them directly into the piece, what would the tools be like, and how would his architectural spaces be different? The Xenakis inspired Reflection Visualizer is an abstract software tool for experimenting with architectural acoustic lenses. It is intended more as an experiment for architectural or musical brainstorming, than as a simulation for analysis of sound propagation. For example:

1. It illustrates sound projection in only two dimensions.
2. It is frequency independent. Real surfaces reflect only wavelengths much smaller than the size of the reflector.³
3. Diffraction is ignored.
4. Acoustic sound waves of higher frequencies propagate more directionally than lower frequencies. This property is ignored.

4.1 Implementation

The Reflection Visualizer was implemented as a web app using the HTML5 Paper.js⁴ vector graphics library. Try Reflection Visualizer online at

<http://web.media.mit.edu/~holbrow/mas/reflections/> Click and drag on any black dot to move the object. Black dots connected by grey lines are handles that re-orient (instead of move) objects.

¹ S. Gradstein, editor. *The Philips Pavilion at the 1958 Brussels World Fair*, volume 20. Philips Technical Review, 1959

² Martijn Vercammen. The reflected sound field by curved surfaces. *The Journal of the Acoustical Society of America*, 123(5):3442, 2008. ISSN 00014966. DOI: 10.1121/1.2934246

³ Zhixin Chen and Robert C. Maher. Parabolic Dish Microphone System, 2005. URL http://www.coe.montana.edu/ee/rmaher/publications/maher_aac_0805.pdf

⁴ <http://paperjs.org/>

On reflection surfaces, the handles adjust the angle and shape of the surface curve. Handles connected to sound sources adjust the angle and length of the sound beams.

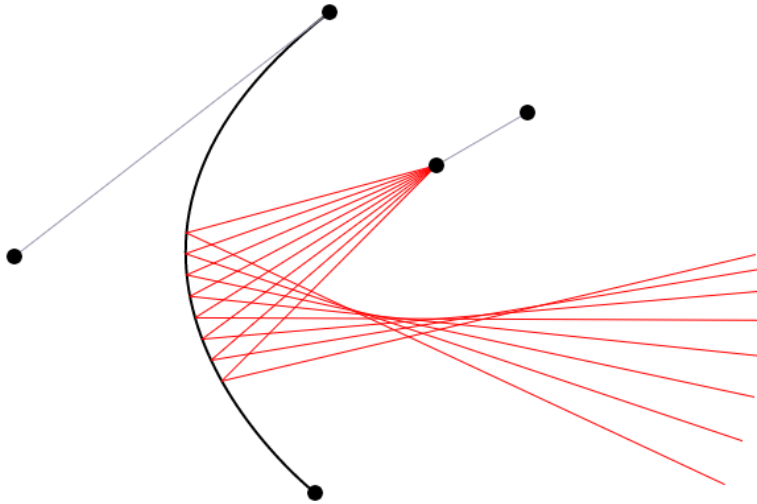


Figure 4.1: Reflection Visualizer user interface.

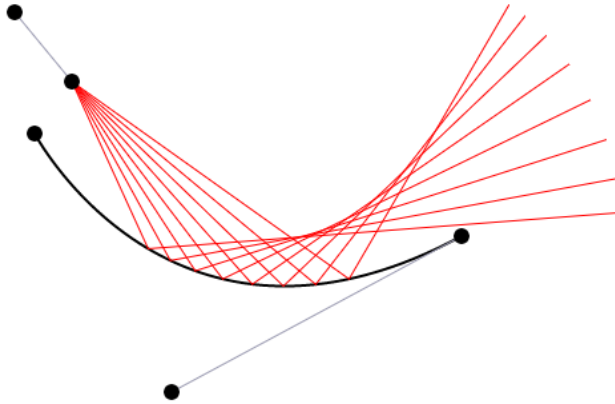
4.2 *Reflection Visualizer Architectural Example*

Assume we are creating the floor plan for a new architectural space and accompanying electronic music performance. The music piece incorporates spatial features in the style of *SOUND=SPACE* by Rolf Gehlhaar: Our audience moves through the performance space, and as they move, the sound changes, making the music experience unique to every visitor. We would like to use acoustic reflections to manipulate the sound in space, such that at certain points the sound is focussed on the listener. When we hear an acoustic sound reflection off a concave surface, the sound can arrive at our ears in two possible states:

1. The path of the sound from the source to the reflecting surface to our ears is equidistant for each point on the reflecting surface. Ignoring any direct sound, the reflection arrives in phase, and the surface acts as acoustic amplifier of the reflection.
2. The path of the sound from the source to the reflecting surface to our ears is slightly different for each point on the surface. All the reflections arrive out of phase with each other.

We can use the Reflection Visualizer tool to prototype potential layouts and gain some intuition about our focal points. The curved

Figure 4.2: Reflections from a 30° loudspeaker arriving out of phase.



black line in the user interface (figure 4.1) represents a reflective surface. The black dot with emanating red lines represents a sound source and the red line represent sound propagation. Each red line emanating from a sound source is the same length, no matter how many times it has been reflected. If it is possible to adjust the length of the red lines such that each one ends at the same spot, it shows that reflections will arrive at that spot in phase. Figures 4.2 and 4.3 show how we can adjust the curve of a surface to focus reflections on a point.

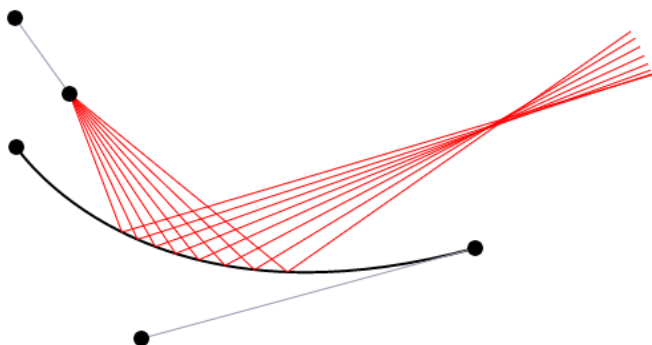


Figure 4.3: By adjusting the curvature of the reflective surface, we can focus the audio reflections.

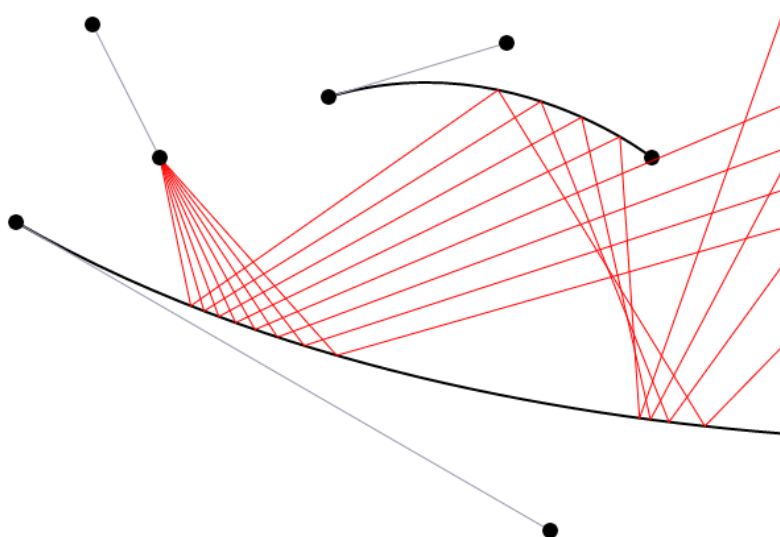


Figure 4.4: A musical composition. The red emanating lines can also be thought of as stochastic pitch swarms, similar to those Xenakis wrote for *Metastasis* in 1954 (figure 2.5).

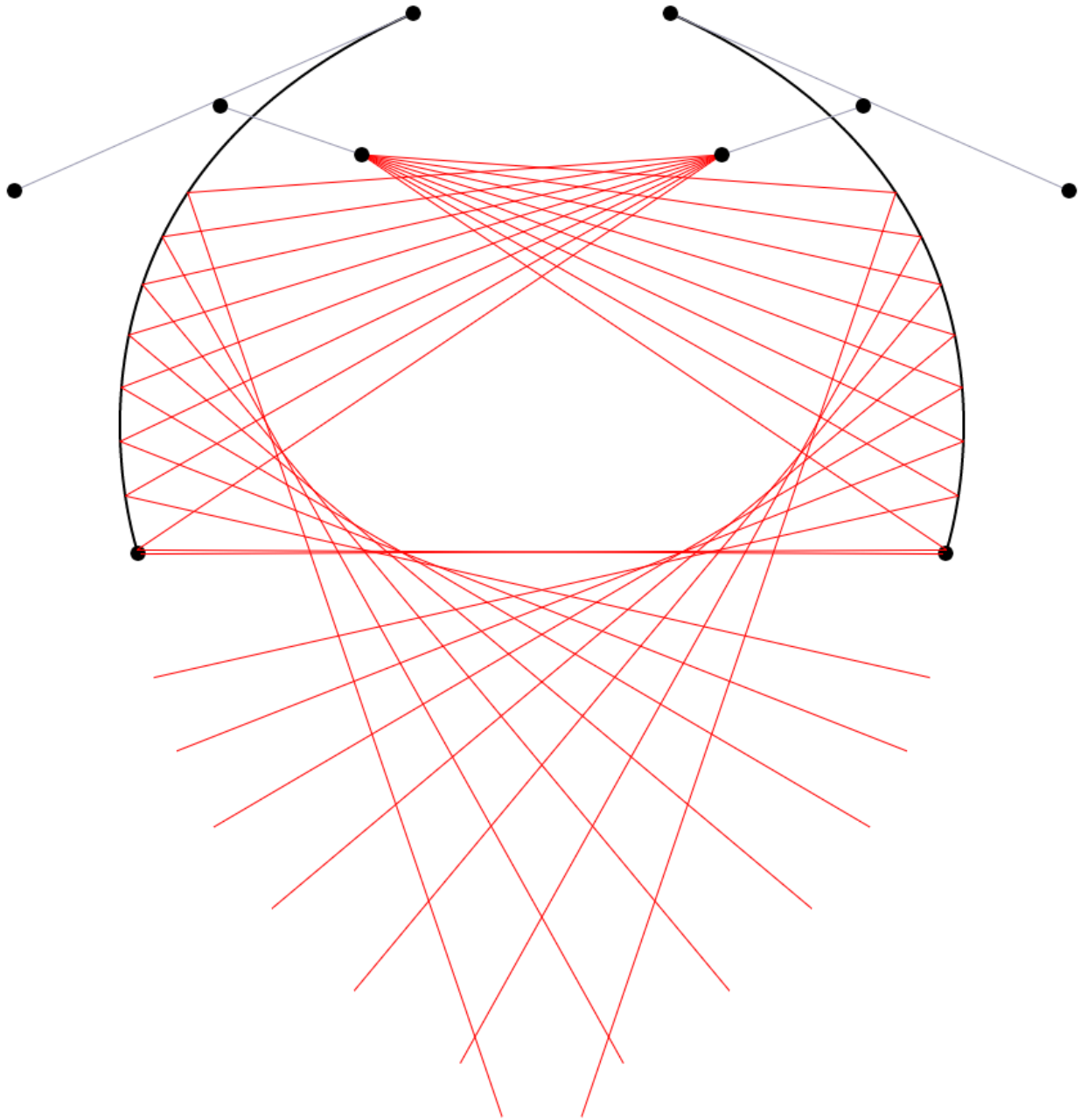


Figure 4.5: The Reflection Visualizer.

5

The Hypercompressor

The inspiration for the Hypercompressor came during the development of *Vocal Vibrations*, an interactive music installation about the human voice and about engaging the public in singing.¹ The project featured a Music Concrète composition, *The Chapel* by Tod Machover, which was mixed in a 10-channel surround sound format and played throughout the installation. During the mixing process, I discovered an important surround sound tool missing from my mixing workflow. When mixing in mono or stereo, audio compression lets us meticulously shape and balance sounds in time. I found myself wishing I could shape and position sounds in space just as easily.

5.1 *Building on the Compression Paradigm*

The design, implementation, and use of traditional dynamic range compression is well-documented in the literature,² so we will describe dynamic range compression only to the extent that needed to explain the foundation for Hypercompression. Imagine we are mixing a vocal pop performance, and during the verse our vocalist sings moderately loud, or *mezzo-forte*. At the beginning of the chorus, our singer wants a full and powerful sound, so she adjusts the dynamic to very loud, or *fortissimo*; however, the new louder dynamic interrupts the balance between the vocals and the other instruments in our mix. We like the powerful sound of our singer's *fortissimo* performance, but our balance would be improved if we had the volume of a *forte* performance instead. One option is to manually turn down the vocalist during the chorus, which in some cases this is the best solution. When we want more precise control, we can use a compressor.

¹ Charles Holbrow, Elena Jessop, and Rebecca Kleinberger. Vocal Vibrations: A Multisensory Experience of the Voice. *Proceedings of the International Conference on New Interfaces for Musical Expression*, pages 431–434, 2014. URL http://www.nime.org/proceedings/2014/nime2014_378.pdf

Unless noted otherwise, “compression” is used in this thesis to describe dynamic range compression, as opposed to data compression.

² Dimitrios Giannoulis, Michael Massberg, and Joshua D Reiss. Digital Dynamic Range Compressor Design - A Tutorial and Analysis. *Journal of the Audio Engineering Society*, 60(6):399–408, 2012; Alex Case. *Sound FX: Unlocking the Creative Potential of Recording Studio Effects*. Focal Press, 2007. ISBN 978-0240520322; and Emmanuel Deruty, Francois Pachet, and Pierre Roy. Human-Made Rock Mixes Feature Tight Relations. *Journal of the Audio Engineering Society*, 62(10), 2014

Traditional Compression

A compressor is essentially an automated dynamic volume control. Most compressors include at least four basic parameters in the user interface that allow us to customize its behavior: *threshold*, *ratio*, *attack time*, and *release time*. We can send our vocalist's audio signal through a compressor, and whenever her voice exceeds the gain level set by our threshold parameter, the signal is automatically attenuated. As the input signal further exceeds the threshold level, the output is further attenuated relative to the input signal. The ratio parameter determines the relationship between the input level and output level as shown in figure 5.1.

Threshold and ratio settings are essential for controlling dynamic range, but the power and creative flexibility of the compressor comes with the attack time and release time parameters. These parameters determine the speed at which the compressor attenuates (attack time) and disengages (release time) when the input signal exceeds the threshold. By adjusting the attack and release times, we can change the temporal focus of the compressor. Consider the following examples:

- Perhaps we want the compressor to engage or disengage at the time scale of a musical phrase. We could set our attack time long enough to let transients through without engaging the compressor significantly (try 20 milliseconds). If our release time is quite long (try 300 milliseconds), and we set our threshold and ratio carefully, we might be able to convince the compressor to smooth musical phrases.
- If we want our compressor to focus on syllables instead of phrases, we can shorten our attack and release times (try 10 milliseconds and 40 milliseconds respectively). When the compressor engages and disengages at each syllable, it imparts a different quality (sometimes described as “punch”).
- If we reduce our attack and release parameters enough, we can instruct our compressor to engage and disengage at the time scale of an audio waveform, compressing individual cycles. This will distort an audio signal, adding odd order harmonics,³ and imparting an entirely different quality.

The attack and release times listed here are a rough guide only. The exact function of these parameters varies from one model of compressor to another, and results also depend on the audio input material, as well as the threshold and ratio settings. The results of audio compression can sometimes be characterized better by a feeling than a formula.

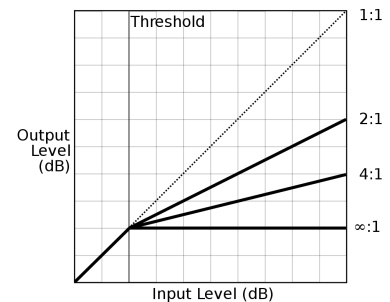


Figure 5.1: “Compression ratio” by Iain Fergusson. Licensed under Public Domain via Wikimedia Commons https://commons.wikimedia.org/wiki/File:Compression_ratio.svg#/media/File:Compression_ratio.svg

³ Not every compressor model can react quickly enough to distort a waveform. The Dbx 160 and Teletronix LA2A are known to be fast enough to distort.

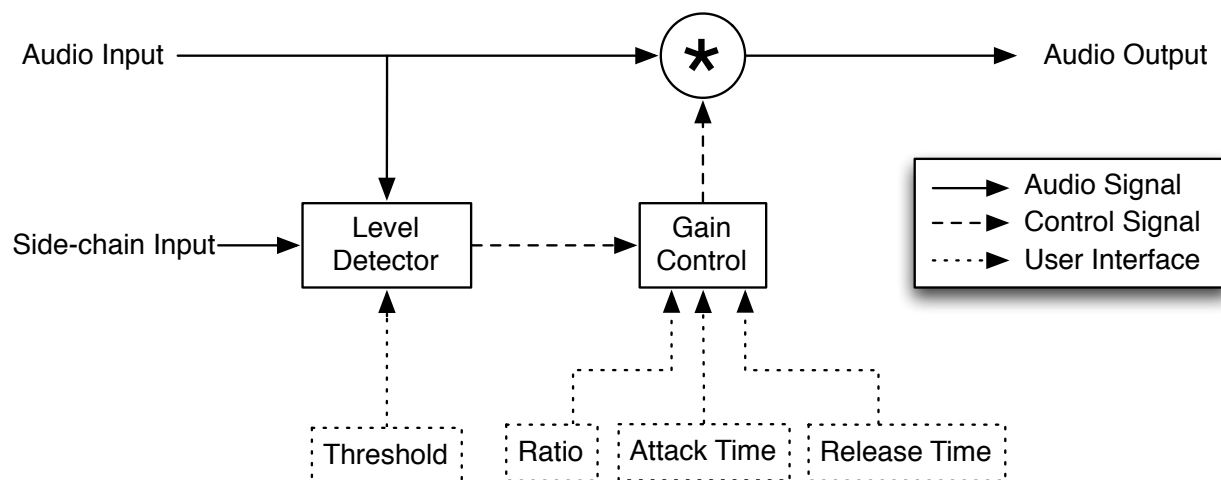


Figure 5.2: Block diagram of a simple monophonic dynamic range compressor.

Side-Chain Compression

Compressors often have an additional operational mode that is the primary inspiration for Hypercompression. As previously discussed, traditional compressors automatically reduce the gain of a signal that exceeds a given threshold. Some compressors allow us to attenuate the level of a signal when a *different* signal exceeds the threshold level. Models that supports side-chain compression have a second audio input. When we switch the compressor into side-chain mode, the compressor attenuates the first signal only when the second signal exceeds the threshold.

Side-chain compression is often used to moderate the balance of kick drum and bass guitar. If the bass guitar is briefly attenuated just enough each time the kick drum hits, we can set the kick and bass guitar at exactly the gain levels we want without one masking the other. Because the bass guitar is only briefly attenuated, it will not be perceived as any quieter. In this example, we use the kick drum to create a gain envelope for our bass guitar. The kick *pushes* the bass to make room for itself. The attack time and release time parameters give control over this behavior in the temporal domain. The next step is to expand this model to add control in the spatial domain.

5.2 Ambisonics

Ambisonics is a technique for encoding and decoding three-dimensional surround sound audio.⁴ Ambisonic audio differs from discrete-channel surround sound formats such as 5.1 and

⁴ Michael Gerzon. Periphony: With-Height Sound Reproduction. *Journal of the Audio Engineering Society*, 21(1):2–10, 1973; and Michael Gerzon. Ambisonics in Multichannel Broadcasting and Video. *Journal of the Audio Engineering Society*, 33:859–871, 1985. ISSN 00047554

7.1, in that it does not depend on a particular speaker configuration. An ambisonic recording can be decoded on many different surround speaker configurations without disarranging the spatial contents of the audio recording.

Imagine we use an omnidirectional microphone to record an acoustic instrument at a sample rate of 44.1 kHz. We sample and record 44100 samples every second that represent the air pressure at the microphone capsule during the recording. Our omni-directional microphone is designed to treat sound arriving from all angles equally. Consequently, all acoustical directional information is lost in the process.

If we want to encode, decode, transmit, or play audio that preserves full sphere 360 degree information, ambisonics offers a solution. Ambisonic audio uses *spherical harmonics* to encode surround sound audio that preserves the direction-of-arrival information that discrete channel recordings (such as mono and stereo) cannot fully capture.

Spherical Harmonics

We know that we can construct any monophonic audio waveform by summing a (possibly infinite) number of harmonic sine waves (Fourier series).⁵ For example, by summing odd *order* sine harmonics at a given frequency f , $(1f, 3f, 5f, 7f, \dots)$, we generate a square wave with fundamental frequency f . As the order increases, so does the temporal resolution of our square wave.

⁵ An excellent description of the transformation between the time domain and frequency domain can be found at <http://betterexplained.com/articles/an-interactive-guide-to-the-fourier-transform/>

By summing sinusoidal harmonics, we can generate any continuous waveform defined in two dimensions (one input parameter and one output). Similarly, by summing *spherical harmonics*, we can generate any continuous shape defined over the surface of a three-dimensional sphere (two input parameters, or polar angles, and one output). Where a traditional monophonic audio encoding might save one sample 44100 times per second, an ambisonic encoding would save one sample *for each spherical harmonic* 44100 times per second. This way we capture a three-dimensional sound image at each audio sample. The number of spherical harmonics we encode is determined by our *ambisonic order*. As our ambisonic order increases, so does the angular resolution of our result on the surface of the sphere.

Spherical Harmonic Definition

For encoding and decoding ambisonics, the convention is to use the real portion of spherical harmonics as defined in equation 5.1, where:

- $Y_n^m(\varphi, \vartheta)$ is a spherical harmonic that is:
 - of order, n
 - of degree, m
 - defined over polar angles (φ, ϑ)
- $N_n^{|m|}$ is a normalization factor.⁶
- $P_n^{|m|}$ is the associated Legendre function of order n and degree m .

$$Y_n^m(\varphi, \vartheta) = N_n^{|m|} P_n^{|m|}(\sin \vartheta) \begin{cases} \sin |m| \varphi, & \text{for } m < 0 \\ \cos |m| \varphi, & \text{for } m \geq 0 \end{cases} \quad (5.1)$$

Given equation 5.1, we can define an ambisonic audio recording as:

$$f(\varphi, \vartheta, t) = \sum_{n=0}^N \sum_{m=-n}^n Y_n^m(\varphi, \vartheta) \phi_{nm}(t) \quad (5.2)$$

Where:

- φ and ϑ describe the polar angle of sound arrival in two dimensions.⁷
- t is time
- $\phi_{nm}(t)$ are our *expansion coefficients*, described below.

Some literature on spherical harmonics swaps the names of *order* and *degree*. In this thesis we use Y_{order}^{degree} . In literature where Y_{degree}^{order} is used, the function of the subscript and superscript remain unchanged; only the names are inconsistent.

⁶ In ambisonic literature (and software), there are multiple incompatible conventions for the normalization of spherical harmonics. The Hypercompressor uses the *Furse-Malham* (FuMa) normalization convention.

⁷ Note that ambisonics uses polar angles to describe the angle of arrival of sound. These are similar to spherical coordinates, minus the inclusion of *radial distance*. Distance is not part of the ambisonic specification.

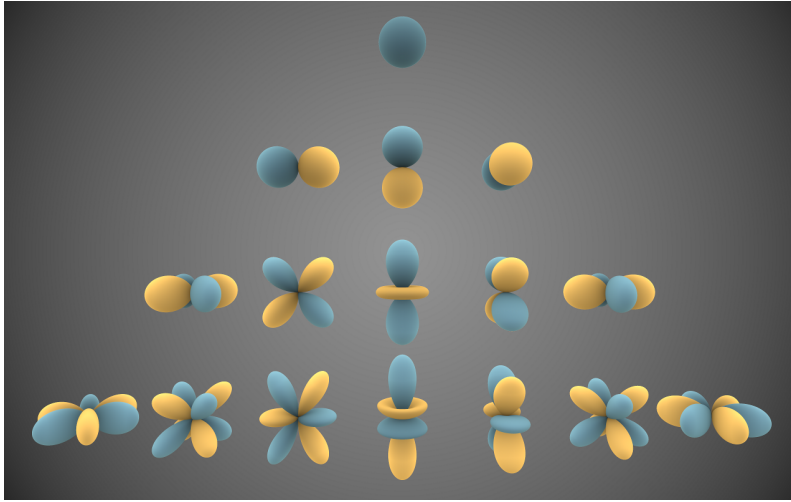


Figure 5.3: Spherical harmonics 0th order (top row) through 3rd order (bottom row). This figure shows the output of $Y_n^m(\varphi, \vartheta)$ for $n = 0, n = 1, n = 2$, and $n = 3$. The distance of the surface from the origin shows the value at that angle. Darker blue regions are positive, while lighter yellow regions are negative. Image credit: Ingo Quilez, licensed under *Creative Commons Attribution-Share Alike 3.0 Unported*.

Spherical Harmonic Expansion Coefficients

In our monophonic recording example, we save just one digital sample 44100 times per second, with each saved value representing the air pressure at a point in time. We know that by summing the correct combination of spherical harmonics, we can describe any continuous function over the surface of a sphere. Instead of sampling air pressure directly, we sample a coefficient describing the weighting of each spherical harmonic 44100 times per second. The resulting sphere encodes the pressure including the direction of arrival information. The weighting coefficients or *expansion coefficients* are recorded in our audio file instead of values representing air pressure directly. Now, by summing together our weighted spherical harmonics, we can reconstruct the fluctuations in pressure, including the angle of arrival information. We can recall this snapshot of information at our 44.1 kHz audio sample rate.

Ambisonic Encoding

There are two ways to create an ambisonic recording. First, we can use a soundfield microphone to record an acoustic soundfield. Soundfield microphones, like the one developed by Calrec Audio, can capture angle of arrival information with the spatial resolution of first order ambisonics.⁸ Alternatively, we can algorithmically encode pre-recorded sources, creating virtual sources in an ambisonic bus.⁹

⁸ Ken Ferrar. Soundfield Microphone: Design and development of microphone and control unit, 1979. URL <http://www.ai.sri.com/ajh/ambisonics/wireless-world-farrar-10-1979.pdf>

⁹ D G Malham and A Myatt. 3-D sound spatialization using ambisonic techniques. *Computer music journal*, 19(4):58–70, 1995. ISSN 0148-9267. DOI: 10.2307/3680991

5.3 *Ambisonic Conventions used for Hypercompression*

This thesis follows ambisonic convention for describing axis of rotation: The x-axis points forward, the y-axis point left, and the z-axis points up. Polar angles are used to describe orientation with 0° azimuth being forward, and increasing as we move to the right. 0° elevation also points forward and increases as we move upward, with 90° being straight up along the z-axis. When working with ambisonics, multiple incompatible conventions exist for ordering and normalizing spherical harmonics.¹⁰ The Hypercompressor uses *Furse-Malham* normalization (FuMa)¹¹, and first order ambisonics with *B-format*¹² channel ordering. B-format ordering labels the four first order ambisonic channels as W, X, Y, and Z, with W being the spherical harmonic of order zero and degree zero, and X, Y, and Z being the pressure gradient components along their respective axes.

¹⁰ Christian Nachbar, Franz Zotter, Etienne Deleflie, and Alois Sontacchi. AMBIX - A Suggested Ambisonics Format. In *Ambisonics Symposium 2011*, 2011

¹¹ D G Malham. Higher order Ambisonic systems, 2003. URL http://www.york.ac.uk/inst/mustech/3d_audio/higher_order_ambisonics.pdf

¹² Florian Hollerweger. An Introduction to Higher Order Ambisonic, 2008. URL <http://flo.mur.at/writings/HOA-intro.pdf>

5.4 Hypercompressor Design

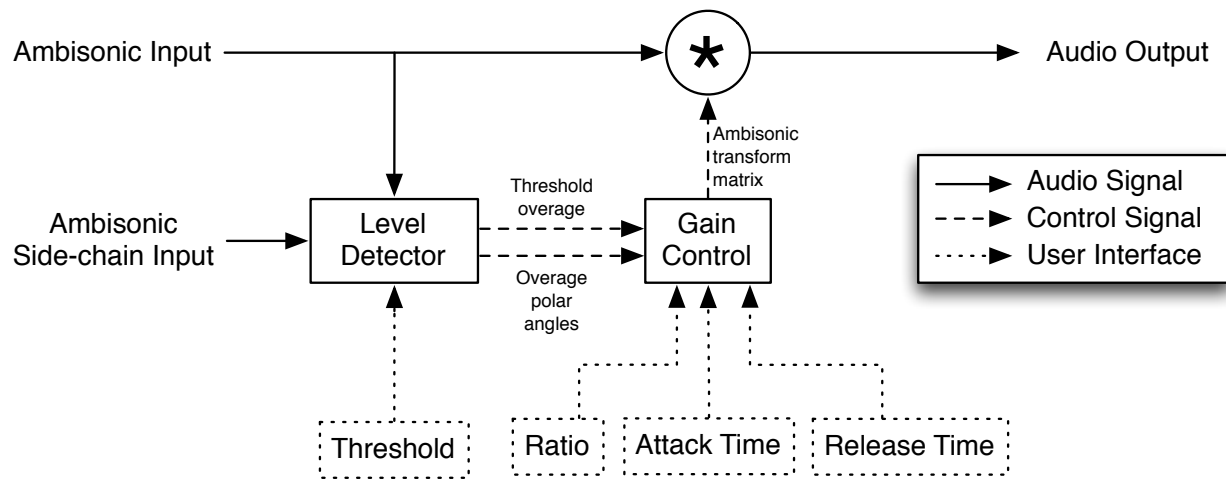


Figure 5.4: Hypercompressor block diagram

The Hypercompressor (or ambisonic compressor) combines the traditional model of compression with the surround sound capability of ambisonics. Given ambisonic input, and an optional ambisonic side-chain input, the ambisonic compressor is intended to process input material in one of two modes:

1. Standard mode: We set a compression threshold, similar to on a traditional compressor. When a region in our surround sound input material exceeds the set threshold, the compressor engages and attenuates only that region.
2. Side-chain mode: This mode takes advantage of a second ambisonic input to our signal processor. When the gain of spatial region in our secondary input exceeds our threshold, we attenuate that same region in the the main input, and output the results.

In both modes, our ambisonic compressor must attenuate and then release attenuation according to the attack time and release time parameters. The block diagram for the Hypercompressor (figure 5.4) can remain mostly unchanged from the the block diagram for our traditional compressor in figure 5.2. The most important changes are:

- Our audio signals must be updated to handle encoded ambisonics. This is as simple as increasing the number of channels on each solid black connection in figure 5.2. The hypercompressor works with first order ambisonics, so every audio path must carry four audio channels.

- On a traditional compressor, the level detector only needs to detect the difference between the gain of the input signal and the gain specified by the threshold parameter. Our ambisonic level detector needs to decode the incoming signals and identify both a threshold overage and the region where the overage occurred.
- Our gain control module needs to listen to the input coming from the level detector module and be able to attenuate the specific regions that exceed our threshold parameter.

Level Detection Module

In *Spatial Transformations for the Alteration of Ambisonic Recordings*, Matthias Kronlachner describes one approach for making a visual ambisonic level meter:¹³

1. Choose a series of discrete points distributed on the surface of a sphere. Ideally the points are equally distributed, so the vertices of platonic solid shapes like the dodecahedron (12-sided polyhedron) and icosahedron (20-sided polyhedron, figure 5.5) work well. For spatial accuracy, Kronlachner recommends a spherical t -design with 240 points described by Hardin and Sloane.¹⁴
2. Evaluate each spherical harmonic at every point chosen. Cache the results in a matrix.
3. With the cached spherical harmonics, it is then possible to calculate the root mean square (RMS) and peak values more efficiently at the audio rate.
4. A level meter does not need to refresh the display at the audio sample rate, so it is acceptable to interpolate between the points on the sphere and update the graphical representation at the control rate, which could be as slow as 30 Hz (approximately every 33 milliseconds).

A similar approach can be used to make an ambisonic level detector; however, a compressor needs to react much more quickly than a level meter. The compressor cannot even *begin* to engage until the level meter has responded, and attack times faster than 33 milliseconds are common in conventional compression. Every point on the sphere requires a buffer to calculate the RMS. We also need to decode ambisonics at the audio sample rate and keep track of peak values. Ideally we would also interpolate between the points.

¹³ Matthias Kronlachner. *Spatial Transformations for the Alteration of Ambisonic Recordings*. Master's thesis, Graz University of Technology, 2014a

¹⁴ R. Hardin and N. Sloane. McLaren's Improved Snub Cube and Other New Spherical Designs in Three Dimensions. *Discrete Computational Geometry*, 15: 429–441, 1996

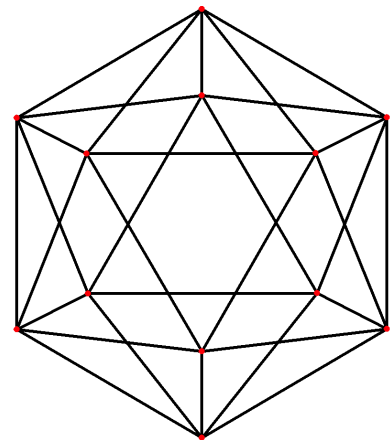


Figure 5.5: An icosahedron

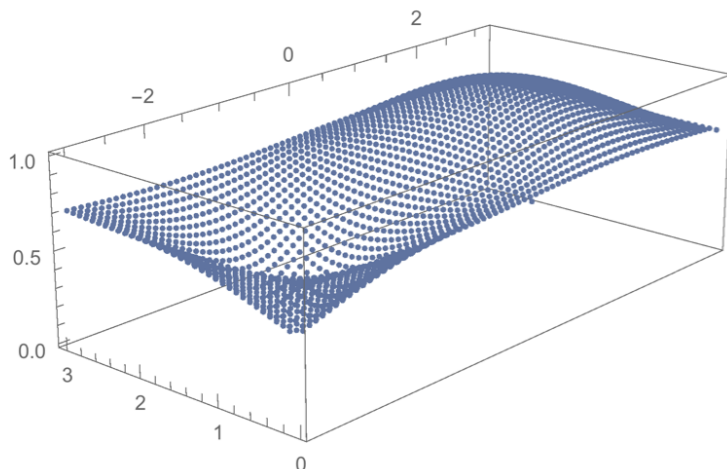


Figure 5.6: Calculation of the cylindrical projection of a single ambisonic panned source in the Wolfram Mathematica software package

An Efficient Level Detection Module

The Hypercompressor needs to detect the level of our ambisonic input material and identify (as quickly as possible) when and where the signal exceeds the compressor threshold. In the interest of computational efficiency, the first level detector I wrote attempted to extract overage information with minimal ambisonic decoding and signal processing.

1. In this level detector, we calculate the RMS average at the center of six lobes corresponding to the first order spherical harmonics: front, rear, left, right, top, and bottom.
2. Calculate a map of the influence of each lobe on the surround sound image (figures 5.6, 5.7). For example, pan a monophonic sound directly forward in an ambisonic mix, cache an image of the resulting sound sphere. Save one image for each of the six lobes.
3. We have six images, each representing one of the six lobes of our first order ambisonic spherical harmonics. In step 1, we calculated the RMS level at each of the corresponding points on our surround sphere. Use the six RMS levels to weight each of our six maps. The sum of the weighted maps shows the gain distributed across our ambisonic sphere.

Ambisonic Efficient Level Detection Module Results If the input to the level detector is encoded as an ambisonic plane wave, this level detector does yield accurate results. In the more common case,

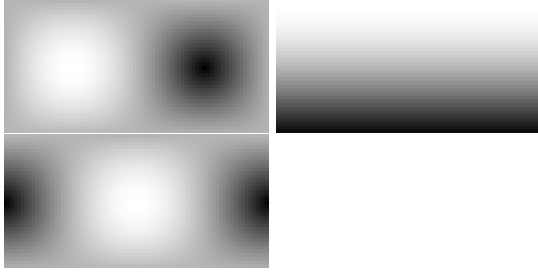


Figure 5.7: Influence maps of three first order spherical harmonics: left, top, and front. Pure white is -0 dBFS, black is $-\infty$ dBFS. Cylindrical projection.

when our ambisonic input material contains multiple sources that are each ambisonically panned to different positions, this interpolation technique does not accurately calculate the RMS at any angle. In simple cases, where we can be sure our input material is appropriate, the technique described here might be useful. For greater spatial resolution (at the expense of performance) the approach described in 5.4 will be more effective.

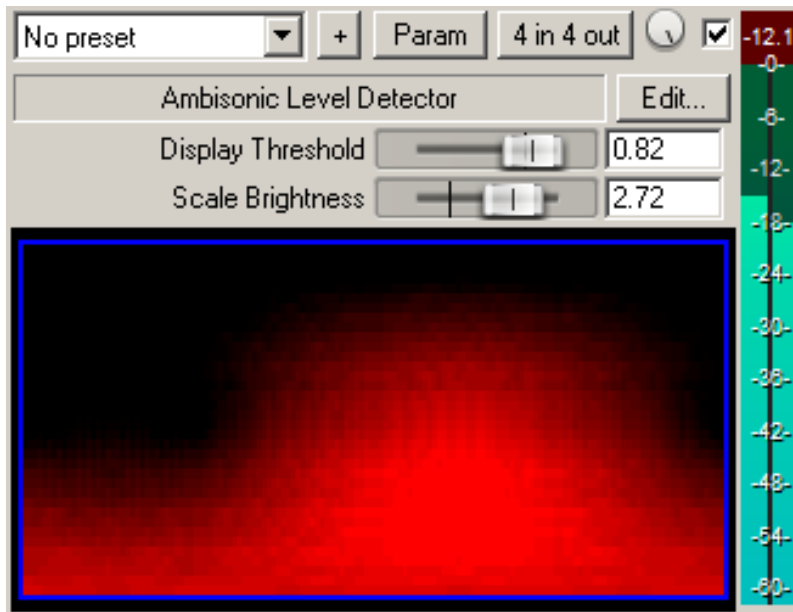


Figure 5.8: The Hypercompressor visualizer written for the efficient ambisonic level detector. The surround sphere is projected to a cylinder and unwrapped on the flat surface. In this image, a monophonic source is panned slightly down and to the right (45° azimuth, -45° elevation).

Ambisonic Gain Control Module

The spherical harmonics defined in equation 5.2 form a set of orthogonal basis functions. If we define a sequence for our spherical harmonics and spherical harmonic expansion coefficients, we can treat a set of expansion coefficients as a vector, and perform matrix operations on them that rotate, warp, and re-orient our three-dimensional surround sound image.¹⁵ The ability to mathemat-

¹⁵ Hannes Pomberger and Franz Zotter. Warping of 3D Ambisonic Recordings. *International Symposium on Ambisonics and Spherical Acoustics*, 3, 2011

ically warp and manipulate our surround sound image makes ambisonics the perfect choice for implementing a surround sound compressor.

The Focus Transform One transform that lets us attenuate a region of the surround sound sphere is the *focus* transform distributed as part of the open source Ambisonic Toolkit (ATK).^{16,17}

$$F(w) = \begin{pmatrix} \frac{1}{1+\sin|w|} & \frac{1}{\sqrt{2}} \frac{\sin(w)}{1+\sin|w|} & 0 & 0 \\ \sqrt{2} \frac{\sin(w)}{1+\sin|w|} & \frac{1}{1+\sin|w|} & 0 & 0 \\ 0 & 0 & \frac{\cos(w)}{1+\sin|w|} & 0 \\ 0 & 0 & 0 & \frac{\cos(w)}{1+\sin|w|} \end{pmatrix} \quad (5.3)$$

¹⁶ <http://www.ambisonictoolkit.net/>

¹⁷ Joseph Anderson. Introducing... the Ambisonic Toolkit. In *Ambisonics Symposium*, 2009

This transform is intended to focus attention on the region directly in front of the listener (0° azimuth, 0° elevation), by attenuating the region in the opposite direction and gently warping the surround sound image toward the front. w is a value between 0 and $\frac{\pi}{2}$ radians and specifies the intensity of the transformation. When $w = 0$, the surround field is unchanged. When $w = \frac{\pi}{2}$ sounds panned hard to the rear are muted, sounds panned to the left and right are attenuated by 6dB, the entire surround sound image is warped to the front, and the gain facing forward is unchanged. This enables us to push one sound out of the way in order to make room for another sound, as described in section 5.4.

Equation 5.3 attenuates the region behind the listener. If we want to attenuate a region other than the rear, we can rotate F using a rotation matrix like the one below.

$$R_z(\varphi) = \begin{pmatrix} 1 & 0 & 0 & 0 \\ 0 & \cos(\varphi) & \sin(\varphi) & 0 \\ 0 & -\sin(\varphi) & \cos(\varphi) & 0 \\ 0 & 0 & 0 & 0 \end{pmatrix} \quad (5.4)$$

Equation 5.4 (from the ATK) describes a rotation around the z-axis, by φ radians. To rotate the focus transform to the right instead of to the front, we first apply the focus transform to the inverse of a 90° right rotation. Then we apply the 90° matrix to the result. This example is generalized by:

$$X(w, \varphi, \vartheta) = R_z(\varphi) R_y(\vartheta) F(w) R_y^{-1}(\vartheta) R_z^{-1}(\varphi) \quad (5.5)$$

Equation 5.5 lets us programmatically generate an ambisonic focus transform matrix that targets a specified region of the surround field, fulfilling the objectives for our ambisonic gain control module in the Hypercompressor.

Ambisonic Gain Control Module Results The focus transform lets us warp the surround field, pushing the field to make room for new sounds. In some cases (for example, when mastering an ambisonic recording), warping the surround sound image is undesirable, and a simple directional gain transform should be used instead (an appropriate transform is defined elsewhere¹⁸). However, the goal of the Hypercompressor is not to compress dynamic range like a traditional compressor. The goal is to compress *space*. The focus transform is a compromise: We partly attenuate a region, but we also bend the surround sound image so that important parts of our surround texture are panned to a position with fewer competing sounds. This is an effect that is not possible with a traditional compressor.

The focus transform also ties attenuation amount to attenuation radius. If we use only a single focus transform, it is not possible to only slightly attenuate a large region of the surround sound field. The following chapter describes how we used this to our advantage during the live performance of *De L'Expérience*.

¹⁸ Matthias Kronlachner. Warping and Directional Loudness Manipulation Tools for Ambisonics, 2014b. URL http://www.matthiaskronlachner.com/wp-content/uploads/2013/01/EAA_2014_Kronlachner_Zotter.pdf

6

De L'Expérience

De L'Expérience is a composition by Tod Machover in eight sections for narrator, organ, and electronics. The piece was commissioned by the *Orchestre Symphonique de Montréal* (OSM) and premiered at the *Maison Symphonique de Montréal* on May 16th, 2015. The text for the piece was taken from the writings of Michel de Montaigne, the 16th century philosopher known for popularizing the essay form. Performers included Jean-Willy Kunz, organist in residence with the OSM, and narrator Gilles Renaud. A recording of the performance has been made available online.¹

¹ http://web.media.mit.edu/~holbrow/mas/TodMachover_OfExperience_Premier.wav

The Organ

The live performance of *De L'Expérience* presented a unique challenge that fits well with the themes in this thesis. The acoustic pipe organ can project sound into space unlike any array of loudspeakers. This is especially true for an instrument as large and magnificent as the Pierre Béique Organ in the OSM concert hall, which has 6489 pipes and extends to approximately 10 meters above the stage. Our objective is to blend the sound of the organ with the sound of electronics.

6.1 *Electronics*

The electronics in the piece included a mix of synthesizers, pre-recorded acoustic cello, and other processed material from acoustic and electronic sources, all composed by Tod Machover. Prior to the performance, these sounds were mixed ambisonically:

1. The cello was placed in front, occupying approximately the front hemisphere of our surround sound image.
2. The left and right channel of the electronic swells were panned to the left and right hemispheres. However, by default they

were collapsed to omnidirectional mono (the sound comes from all directions, but has no stereo image). The gain of this synth was mapped to directionality, so when the synth grows louder, the left and right hemisphere become distinct from each other, creating an illusion of the sound is growing larger.

3. Additional sound sources are positioned in space, such that each has as wide an image as possible, but overlaps with others as little as possible.

The overarching goal of this approach was to create a diverse but interesting spatial arrangement, while keeping sounds mostly panned in the same spot: movement comes from the warping of the surround image by the Hypercompressor.

Sound Reinforcement

Loudspeakers were positioned throughout OSM concert the hall. A number of factors went into the arrangement: audience coverage, surround coverage, rigging availability, and setup convenience. All speakers used were by Meyer Sound.² A single CQ-2 was positioned just behind and above the narrator to help localize the image of his voice. JM-1P speakers on stage left and stage right were also used for the voice of the narrator, and incorporated into the ambisonic playback system. Ten pairs of UPJ-1Ps were placed in the hall, filling in the sides and rear for ambisonic playback, two at the back of the hall, mirroring the CQ-2s on stage, four on each of the first and third balconies. The hall features variable acoustics, and curtains can be drawn into the hall to increase acoustic absorption and decrease reverb time. These were partially engaged, striking a balance: The reduced reverb time improved the clarity of amplified voice, while only marginally impacting the beautiful acoustic decay of the organ in the hall. The show was mixed by Ben Bloomberg. Ambisonic playback and multitrack recording of the performance was made possible with the help and expertise of Fabrice Boissin and Julien Boissinot and the Centre for Interdisciplinary Research in Music Media and Technology (CIRMMT) at McGill University.

² <http://www.meyersound.com/>

A note on composition, performance, and engineering No amount of engineering can compensate for poor composition, orchestration, or performance. A skilled engineer with the right tools only can only mitigate shortcomings in a performance. Good engineering starts and ends with good composition, arrangement and performance. I have been quite fortunate that all the musicians involved with *De L'Expérience* at every stage are of the highest caliber.

6.2 *Live Hypercompression Technique*

During the performance, the encoded ambisonic electronic textures were patched into the main input of the Hypercompressor, before being decoded in realtime using the *Rapture3D Advanced* ambisonic decoder by Blue Ripple Sound.³ Four microphones captured the sound of the organ: two inside and two hanging in front. The placement of the mics was intended to capture as much of the sound of the organ as possible, and as little of the sound of the amplified electronics in the hall. These four microphone signals were encoded to ambisonics in realtime, and the resulting ambisonic feed was patched into the side-chain input of the Hypercompressor. In this configuration, the organ drives the spatialization of the electronic sounds. By ambisonically panning the organ microphones, we can control how our electronics are spatialized. After some experimentation we discovered the best way to apply the Hypercompressor in the context of *De L'Expérience*. When the organ was played softly, the sound of the electronics filled the performance hall from all directions. As the organ played louder, the electronic textures dynamically warped toward the organ in the front of the concert hall. The spatial and timbral movement of the electronics together with the magnificent (but stable) sound of the organ created a unique blend that would be inaccessible with acoustic or electronic sounds in isolation.

³ <http://www.blueripplesound.com/products/rapture-3d-advanced>



Figure 6.1: The Pierre Béique Organ in the OSM concert hall during a rehearsal on May 15th, 2015. Approximately 97% of the organs' 6489 pipes are out of sight behind the woodwork. Photo credit: Ben Bloomberg

7

Discussion and Analysis

In the previous chapters we explored three new tools for creating and processing music, including their motivations and implementations. Stochastic Tempo Modulation in [chapter 3](#) proposed a mathematical approach for composing previously inaccessible polytempic music. The Reflection Visualizer in [chapter 4](#) introduced an interface for quickly sketching abstract architectural and musical ideas. Chapters [5](#) and [6](#) described the motivations for an implementation of a new technique for moving music in space and time. Each of these projects builds on Iannis Xenakis' theory of *stochastic music* and incorporates elements from other disciplines, including mathematics, computer science, acoustics, audio engineering and mixing, sound reinforcement, multimedia production, and live performance.

This final chapter discusses how each project succeeded, how each project failed, and how future iterations can benefit from lessons learned during the development process.

7.1 Evaluation Criteria

To evaluate a project of any kind, it is helpful to begin with a purpose, and then determine the granularity and scope of the evaluation.¹ We might evaluate a music recording for audio fidelity, for musical proficiency of the artist, for emotional impact or resonance, for narrative, for technological innovation, for creative vision, or for political and historical insight. Similarly, we can evaluate the suitability of an analog to digital converter (ADC) for a given purpose. If our purpose is music recording, we might prefer different qualities than if our purpose is electrical engineering. A recording engineer might prefer that the device impart a favorable sound, while an acoustician may prefer that the device be as neutral as possible.

In the evaluation of a music recording, and the evaluation of an

¹ Jerome H. Saltzer and M. Frans Kaashoek. *Principles of Computer System Design An Introduction*. Morgan Kaufmann, Burlington, MA, 2009. ISBN 9780123749574

ADC, we concern ourselves with only the highest level interface: When evaluating a music recording, we listen to the sound of the recording, but we do not evaluate the performance of the ADC used to make the recording. Evaluation is simplified when we consider fewer levels of abstraction.

Stochastic music theory is a *vertical integration* of mathematics, the physics of sound, psychoacoustics, and music. The theory of stochastic music begins with the lowest level components of sound and ends with a creative musical product. What is a reasonable perspective from which to evaluate stochastic music? From the perspective of listening or performing the music? From the perspective of a historian, evaluating the environment that led to the composition or studying the impact on music afterwards? Should we try to make sense of the entire technology stack, or try to evaluate every layer of abstraction individually? Somehow, between the low-level elements of sound and a musical composition or performance, we transition from what is numerically quantifiable to what we can only attempt to describe.

In my evaluation, I focus on two qualities. First, I study how each project achieved its original objectives and how it fell short. Second, I consider how each project can influence or inspire future iterations. I avoid comparative analysis or evaluation based on any kind of rubric. Instead, I evaluate the results of the project according to its own motivations and historical precedents.

7.2 Stochastic Tempo Modulation

This chapter presents a very pure and elegant solution to a very complex problem. But is it important? Is it a significant improvement on the existing techniques presented in section 2.2? If a performer cannot play precise tempo curves anyway, what is this actually for?

Western polytempic music as defined in chapter 3 has existed for only slightly over one century, and there is certainly room for new explorations. The oldest example of Western polytempic music is by Charles Ives in his 1906 piece, *Central Park in the Dark*.² In the piece, the string section represents nighttime darkness, while the rest of the orchestra interprets the sounds of Central Park at night. Beginning at measure 64, Ives leaves a note in the score, describing how the orchestra accelerates, while the string section continues at a constant tempo:

From measure 64 on, until the rest of the orchestra has played measure 118, the relation of the string orchestra's measures to those of the other instruments need not and cannot be written down

² John Greschak. Polytempo Music, An Annotated Bibliography, 2003. URL <http://www.greschak.com/polytempo/ptbib.htm>

exactly, as the gradual *accelerando* of all but the strings cannot be played in precisely the same tempi each time.

Ives acknowledges that there is no existing notation to exactly describe the effect that he wants, and that musicians are not capable of playing the transition in a precise way. In this example, it is not important that the simultaneous tempi have a precise rhythmic relationship. Ives' use of parallel tempi is a graceful one. He achieves a particular effect without requiring the musicians to do something as difficult as accelerate and decelerate relative to each other, and then resynchronize at certain points.

All polytempic compositions must grapple with the issue of synchronicity, and many demand more precision than *Central Park in the Dark*. Stockhausen's *Gruppen* uses polytempi very aggressively, going to great lengths to ensure that the three orchestras rhythmically synchronize and desynchronize in just the right way. If Stockhausen had been able to control the synchronicity of the tempo precisely, it seems likely that he would have wanted to try it.

Some music (and perhaps stochastic music in particular) may be more interesting or influential from a theoretical perspective, than for the music itself in isolation. It could be that the possibilities unlocked through the equations derived in [chapter 3](#) are not different enough from the approximations used by Nancarrow and Cage or that it is unrealistic to direct performers to play them accurately enough to perceive the difference.

However it is surprising that current digital tools for composition do not let us even *try* Stochastic Tempo Modulation. If we want to hear what tempo transitions like the ones describe here sound like using digital technology, there is no software that lets us do so, and we are still forced to approximate. Audio programming languages like Max and SuperCollider let us code formulaic tempi into our compositions, but equations like the ones derived here are still required. I could not find any technique that lets us create swarms of tempo accelerations that fit the constraints described in [chapter 3](#), or any musical example that proposed to have found another solution.

For some cases approximation is perfectly acceptable. If a musician is incapable of playing the part, we are also likely incapable of hearing the subtleties that distinguish an approximation from a perfect performance. However, if we want large collections of simultaneous polytempi, like the ones shown in [figures 3.2](#) and [3.3](#), the approximations possible with transcriptions, or the approximations of an unassisted human performers, are not precise enough.

Future Directions Bryn Bliska’s composition (linked in section 3.5) is a good starting point for future explorations, but it was composed with an earlier version of the polytempic equation that did not allow for large swarms of tempi. In its current state, the polytempic work described in this thesis is just a beginning. We have not yet tried to compose a piece that fully incorporates Stochastic Tempo Modulation.

Equations alone do not make a musical instrument, and composition is difficult without a musical interface. There are a few modern examples of polytempic projects (see chapter 3), but I could not find any examples of interfaces for composing with coordinated mass of tempi. The most exciting direction for this project is the creation of new musical interfaces for composing and manipulating stochastic tempo swarms.

7.3 *Reflection Visualizer*

This project provides a single abstract interface that approaches composition of space (architecture) and the composition of music at the same time. The forms that it makes are familiar from the ruled surfaces seen in Xenakis’ compositions and early sketches of the Philips Pavilion. From a musical perspective, we can think of the x and y axes representing time and pitch. From an architectural perspective the canvas might represent the floor plan of spaces we are designing.

While it is interesting to switch our perspective between the two modes, there is not a clear connection from one to the other. A carefully designed surface or reflection in one mode would be quite arbitrary in the other mode. The reason that the interface is capable of working in both modes is because it is so abstract that it does not commit to one or the other. This is not a complete failing: The tool was really designed to be a brainstorming aid at the very beginning of the design process. It can be much simpler and quicker to use than proper architectural software as a means of creating abstract shapes, similar to sketching on paper, before turning to specialized software for more detailed design.

Curves, Constraints, and Simplicity Despite the limitations of this project, the parts that worked well do form a strong base for future iterations. There is something simple and *fun* about the user interface. There is only one input action; dragging a control point. It is immediately clear what each control point does when it is moved. It is easy to not even notice that there are five different types of control points and each has slightly

different behavior. It is very intuitive to adjust a reflection surface such that the red beams *focus* on a certain point, and then re-adjust a reflection surface so that they diverge chaotically. There is something fascinating about how the simple movements intuitively produce coordinated or chaotic stochastic results.

The red “sound lines” have three degrees of freedom: position, direction, and length. We can point the rays in any direction we like, but their movement is somewhat constrained. The projection angle is locked to 30 degrees and the number of beams is always eight, and most of the flexibility from the interface comes from the reflective surfaces.

Stochastic by Default The Reflection Visualizer interface makes it easier to draw a curving reflective surface than a straight one. If you make a special effort, it is possible to make one of the surfaces straight, but just like drawing a line on a paper with a pen, curved surfaces come more naturally. The curves in the Reflection Visualizer do not come naturally because they are following an input gesture like most “drawing” interfaces, but because of the simple mathematics in the of the Bezier curves. If we consider the red lines to be notes on a time/pitch axis, the default interpretation is stochastic glissandi rather than static pitches. Most musical software assumes static pitches by default and most architectural software assumes straight lines.

Future Directions The obvious next steps for this project involve correcting the shortcomings described above. It could be made to work in three dimensions, and model precise propagation of sound rather than a very simplified abstraction: It could become a proper acoustical simulator. Another possibility is turning it into a compositional or performative musical instrument where we can hear the stochastic glissandi in realtime. These options are not necessarily mutually exclusive, but as the interface becomes tailored to a more specific application, our ability to think about the content as abstract representations also breaks down. The ideal of software that is equally well-equipped to compose music and to imagine architectural spaces is probably unrealistic.

Any visual representation of music is quite abstract, and different visual representations can encourage us to think about music in new and unusual ways. For example each red line can be considered pitch, but it can also be considered its own time axis. By calculating the red paths, we can creating many time axes that follow similar but slightly different trajectories. Alternatively, each red line can be thought of as a time axis for an individual

pitch. When the lines collide with a curved surface after slightly different lengths, it represents an arpeggiated chord. In contrast, a non-arpeggiated chord is represented when the red lines all collide with a surface after traveling identical distances. The abstract nature of this interface leaves room for our imagination to interpret unexpected new musical possibilities.

7.4 *Hypercompression*

The design and development of Hypercompression happened in parallel with pre-production for *De L'Expérience*, and the Hypercompressor was, in part, tailored to the needs of a somewhat unique situation. The resulting project leaves significant design questions surrounding ambisonic dynamic range compression unanswered. For example: What is the best way to detect and attenuate a region on our surround sphere that is an unusual or elongated shape? Should the compressor attempt to attenuate the narrow region only? Should we attenuate the center of the region more than the edge?

When a region of our surround sound image exceeds the Hypercompressor's threshold, the compressor warps the surround image in addition to attenuating the region where the threshold overage occurred. This makes sense for side-chain compression, but is less applicable to standard compression. We could have chosen only warping, or only attenuation, each of which represents its own compromise:

- We could simply warp all sounds away from a region that exceeds the compression threshold without attenuating them at all. However, doing so would increase the perceived level of the sound coming from the opposite direction. We also run the risk of creating sonic “ping-pong” of sounds arbitrarily panning. This can sound exciting, but quickly becomes a contrivance or gimmick.
- If we simply attenuate a region that exceeds the threshold, we are not taking advantage of the opportunities provided to us by surround sound in the first place. In side-chain mode, we risk hiding a compressed sound completely when we could simply warp that region of the surround field to a location where it can be heard more clearly.

The current implementation also does not handle the case when two separate regions of the surround field both exceed the threshold.

De L'Expérience The main goal of using the Hypercompressor was to blend the electronic textures with the sound of the Pierre Béique organ in Tod Machover's composition. The chosen approach was to give the electronics a sense of motion that the organ (whose sound is awe-inspiring, but also somewhat static) cannot produce; thus the electronics can be heard moving *around* the sound of the organ, rather than being required to compete with the sound of the organ. The first attempt at this goal, however, did not go as planned.

The electronics were mixed to occupy as much of the surround sound sphere as possible, filling the entire room with sound. My original idea was to spatially separate the organ and electronics by connecting them to the Hypercompressor in side-chain mode. When the organ was playing it would *push* the sound of the electronics to the back of the room, making it easier to hear both timbres without either masking the other. During the *De L'Expérience* rehearsal, this was the first approach I tried, but the resulting surround texture had a different problem: The sound of the organ and the sound of the electronics were *too* separate. They did not blend with each other in space, but existed as two clearly distinct sources. I arrived at the solution described in [chapter 5](#) only after first trying the exact opposite of the final approach. While I had to revise my strategy during the rehearsal, I consider the Hypercompressor to have aided the blending of the organ and electronics especially well. It is important to note that the beautiful blend of sounds that we achieved would not have been possible without many other contributing factors, such as the expert composition of the electronic textures.

Future Directions The next step is to make a fully featured surround compressor with options for warping and attenuation. Parametric control over the width and spatial resolution of the regions to be attenuated could also help turn the Hypercompressor into a general purpose tool that would be useful in a wide variety of different situations. A more creative path would be to add additional directional effects that can be modulated by the gain of audio content in the surround image. For example, when a region of the surround sphere exceeds our threshold, we could apply a sliding delay to that region. We could also offer a selection of different effects that can all be modulated by the positional gain of the sound sphere. We could even add spatial detection of other sound qualities and create a modulation matrix. For example, we might use a high frequency filter to modulate a phasing effect. This would involve detecting the regions of the sound sphere that

have the most high frequency content, and then proportionally applying a phasing effect to those regions. By expanding on the paradigm of compression in these ways, we unlock possibilities for previously unimaginable surround soundscapes.

7.5 *Stochos*

Hypercompression is a complete realization of a musical idea. Beginning with an objective and a mathematical foundation, we designed and built a custom software implementation and applied it in a live performance context. A study of the process has revealed what is probably the greatest strength of stochastic music theory: The vertical integration of the theory of sound and music lets us study music from a privileged perspective, while the controlled chance built into the system helps us to uncover possibilities that could not be found with conventional means. In the case of Hypercompression, we move sounds in space based on matrix transforms, that are themselves driven by the controlled chance of a random performance. The angular position of our sounds are defined by both explicit mathematical formulas and the unpredictable qualities of live performance.

From a broad perspective all three projects emerged “by chance” in the same way. Each one is the result of musical exploration in a space that indiscriminately draws from mathematics, computer science, acoustics, audio engineering and mixing, sound reinforcement, multimedia production, and live performance. By treating all these disciplines as components of music theory, we discover new musical patterns and possibilities for shaping sound in time and space.

Epilogue

In 2004, the Culture 2000 Programme, created by the European Union approved a grant to an Italian multimedia firm for a project called Virtual Electronic Poem (VEP).³ The project proposed the creation of a virtual reality experience in which users could enter a simulated version of the famous Phillips Pavilion. While developing the VEP, the design team went through the archives of Xenakis, Le Corbusier, and Philips, uncovering every relevant bit of information in order to make the experience as real as possible.⁴

Virtual reality technology changed so much between 2004 and 2015 that reviving the VEP project today would likely involve an additional multimedia archeology expedition as intensive as the first: It would probably be easier (and more effective) to start from scratch using the original documentation. A common problem with multimedia performances is that technology changes so fast that it quickly becomes very difficult to restore even moderately recent projects.⁵ In contrast, the mathematical language that Xenakis used to describe his work is as well-established as the language of Western music notation, and for this reason we have a surprisingly thorough understanding of his music today. It is my hope that the documentation in this thesis will provide an equally dependable and enduring a description of the process of modern musical composition.

³ Culture 2000 Programme Results. Technical report, European Union Culture 2000 Programme, 2004. URL <http://ec.europa.eu/culture/tools/documents/culture-2000/2004/heritage.pdf>

⁴ Vincenzo Lombardo, Andrea Valle, John Fitch, Kees Tazelaar, and Stefan Weinzierl. A Virtual-Reality Reconstruction of Poeme Based on Electronique Philological Research. *Computer Music Journal*, 33(2):24–47, 2009

⁵ Vincenzo Lombardo, Andrea Valle, Fabrizio Nunnari, Francesco Giordana, Andrea Arghinetti, Università Torino, and M M Park. Archeology of Multimedia. In *ACM International Conference on Multimedia*, pages 269–278, 2006. ISBN 1595934472

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