# Hearing Images and Seeing Sound:
# The Creation of Sonic Information Through Image Interpolation

Maxwell Dillon Tfirn
Charlottesville, VA

B.M., University of Florida, 2009
M.A., Wesleyan University, 2012

A Dissertation presented to the Graduate Faculty
of the University of Virginia in Candidacy for the Degree of
Doctor of Philosophy

Department of Music

University of Virginia
April, 2017

## Abstract

This dissertation focuses on the compositional use of spectra, from the inception of this approach in the focus on acoustic characteristics of sound to my own approach, using images and raw data as spectral generators. I create these spectra using image interpolation and processing, and map them from the visual to the auditory domain. I situate my work in relation to the French and Romanian spectralists, the German Feedback Group, and other offshoots whose starting point is timbre-based composition. My own suite of software translates images into sound, allows for the creation and deletion of images, thus altering the sound, and supports analysis of the resultant spectra. Finally, the application of my software and techniques is illustrated in three of my compositions: *Establish, Corrupt, Broken* for 8-channel fixed media, *Laniakea Elegant Beauty* for Chamber Ensemble, and *Invisible Signals* for open instrumentation.

**Acknowledgements**

My dearest thanks go to my advisor, Judith Shatin. Her amazing insights and editorial expertise made this dissertation possible. I have enjoyed working with her during my years at UVA on a number of interesting projects and her continued support and critical questions have helped me to grow as a composer and artist. I would also like to thank my other departmental committee members Ted Coffey and Luke Dahl. Their leadership as teachers, artists, and friends has left a lasting impact on my personal and creative life. Thank you to Roseanne Ford for serving as the outside reader on my committee.

Thank you to my parents, Lori and David Tfirn, and my brother Ian for supporting me along the way. Finally, I owe the greatest amount of gratitude to my wife Seung-Hye. I would not have been able to do this without her support, analytical comments, patience and ability to motivate and inspire me.

# Table of Contents

# Chapter I.

The History of Spectral Music

## Introduction

Spectral music evolved from a new way of thinking that focused on acoustic characteristics of sound. This shift to timbre-based composition has many examples in the early twentieth century, including, for example, Béla Bartók's acoustic scale in his *String Quartet No.5*, Henry Cowell's extended piano techniques in *The Banshee*, Arthur Honegger's inspiration from a steam locomotive in *Pacific 231*, Per Nørgård's infinity series in *Voyage into the Golden Screen*, and many more. Composer Francesco Balilla Pratella, in his *Manifesto of Futurist Musicians,* talked of the emerging arts as "…everlastingly renewed by the varied aspects of modern life and its infinity of intimate relationships with nature" (Pratella, 2012, p. 29). While the futurists were interested in noise and experimentation, the timbral focus of their thinking likewise exhibited a shift in musical thought.

One of those who helped shape a new approach to timbre was Jean-Claude Risset. His *Introductory Catalogue to Computerized Sounds* (1969), accompanied by recordings created by additive synthesis (Beauchamp, 1971, p. 348), was revolutionary because it provided an understanding of the science behind sound synthesis. Shortly after, in 1970, Gérard Grisey and Tristan Murail initiated the spectral school (Fineberg, 2000, p. 1). Composers Michaël Lévinas and Hugues DuFourt joined them, eventually forming

*Ensemble I'itinéraire* in 1974 (Anderson, 2001), giving the composers a platform for their musical experimentation.

This first chapter provides any overview, following the evolution of spectral music as well as its changing definition. It also considers the influence of spectral music on other genres of music.

**1.1 The French School**

Gérard Grisey, Tristan Murail, Michaël Lévinas and Hugues Dufourt explored spectral music from both the compositional and aesthetic standpoints in the 1970's. This music differed from that of its predecessors because it was created in relation to specific acoustic properties. It was made possible by the development of analogue sonograms and software, such as Patchwork from IRCAM, in the mid-to-late 1970's. For the first time, composers could visualize sound and utilize concrete scientific information in their work. It also enabled composers to manipulate sound spectra to create new sonorities that evolve over time.

The term 'spectral music' was coined by Hugues Dufourt for music based on spectral analysis. In his article *Musique Spectrale* (1986), he wrote about the structure of spectral music and the relationship between the composer and the timbral qualities of the sound source (Fineberg, 2000c, p. 2). This new compositional practice was based on microstructures of a changing sound and on the use of spectra to shape the harmonic and melodic structure, as well as anchor the orchestration (Cornocello, 2000, p. 2).

Spectralism aimed to move away from methodologies such as those found in dodecaphonic music, and to investigate the structure of sound itself (Mabury, 2006, p. 25). *Musique Spectrale* gave an overview of the new spectral style of composition. However, as explained later, many disagreed with the views put forth there.

According to Viviana Moscovich (1997), "The spectrum or group of spectra replace harmony, melody, rhythm, orchestration and form" (p.22). She later states that because the spectrum of a sound is always moving, it influences the rhythm and formal structure upon which the composition could be based (Moscovich, 1997, p. 22). While Moscovich's second statement accurately touches on the spectral compositional process, one can argue with the first statement. The spectrum of a sound does not replace, but rather can be used to create or enhance, the harmony, melody, rhythm, orchestration and form. Moscovich's statement takes the creativity away from the composer and reduces the process of composing to a manifestation of the spectrum.

Instead, according to Murail, one has to think of the continuum before thinking of the discrete, have a global approach and not a cellular or sequential one, use logarithmic/expositional methods of organization, build in a functional way, and be concerned with the relationships between conception and perception (Moscovich, 1997, 22). Murail's ideas do not always have a direct relationship to musical properties, but rather suggest how to think about them. In essence, Murail wants the composer to think about a range of spectral properties, apply a logic to the whole system, use logarithmic methods, and be concerned with whether or not the underlying concept itself can be perceived. Murail's statements demonstrate that his focus is on the conceptual aspects of spectral composition rather than on the actual sound.

Grisey's ideas about spectral music focus, rather, on the sound itself. He claims that the musical form should be derived from the evolution of the sound and then asks these questions: Where does the sound come from? Where does it go? What is its way between the manifolds? What sense does it have in this place and in that one (Moscovich, 1997, p.25)? Grisey's ideas are more focused on inspecting the sound, understanding the origin and realizing the changes from one point to another. Grisey also preferred the term "liminal music" to "spectral music", as a way of describing this type of music and its processes (Hurel, 2007). According to Hurel (2007) the word 'liminal' is taken from the Latin word *limen* meaning threshold. Grisey wanted to use the word liminal to describe the thresholds between harmonicity and inharmonicity, fusion to distraction and more (Hurel, 2007). The broad and ambiguous questions and statements by Grisey and Murail show that spectral music is not a genre, but rather a practice that can span different styles and practices (Fineberg, 2000c, p.2).

### 1.2 The First Spectral Works

The early spectral works of Grisey and Murail show a contrast in stylistic approach and logic. The focus of this discussion will be on how the spectrum and its intricate parts are represented in representative pieces and how the musical forms reflects the overall spectrum in each case. Because spectral music is based on a spectrum of an acoustic sound or electronically generated sound, notational practices will be discussed to show how different timbral aspects can be represented.

### 1.2.2 Gérard Grisey's *Partiels*

*Partiels* was composed in 1974 and is still considered a prime example of spectral music (Fineberg, 2000a, p. 116). *Partiels* is scored for a chamber ensemble consisting of flute, oboe, clarinet, bass clarinet, trumpet, trombone, accordion or electric organ, percussion, violin, viola, cello and bass. Its design is based on the analysis of a trombone playing E2, and attempts to resynthesize the spectrum using the ensemble while making a more complex spectrum. Instruments are orchestrated to reflect partials, attack, decay and amplitudes of the trombone sound.

Grisey uses the trombone frequencies rounded to the nearest quarter tone in order to stay as close as possible to the original sound. He also uses crescendo and diminuendo markings that have zigzag lines representing rapid fluctuations of dynamic, as seen in Figure 1.1. This allows for intricate dynamic fluctuation taking place in the trombone spectrum to be represented in a way that performers can play. In addition, the trombone spectrum and the attack and decay of the notes do not line up perfectly with musical time. Grisey uses intricate rhythmical markings to indicate where the beat is, how long to hold the sound, when to look at the conductor, improvised time, when to repeat a given passage, and when to keep the sound improvised for the duration of a breath (Grisey, 1980).

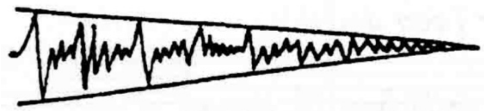


*Figure 1.1 Dynamic Marking from Partiels 1975*

In addition to matching the complex rhythm of the trombone sample, Grisey also incorporates timbral differences. To do so, he uses symbols for a variety of techniques, including normal flutter tonguing, flutter from the throat, breath noise, forced release of the lips and multiphonics to name a few (Grisey, 1980). By using these techniques, Grisey explores acoustic instrument sounds as a means to represent certain aspects of the spectrum. His notational system shows control over parameters of rhythm and pitch that fall outside of traditional practices, and include an element of imprecision.

The instruments rarely play traditionally notated rhythms. Instead Grisey uses slashes through beamed notes indicating to play as fast as possible, or he uses feathered beaming, to show the gradual speeding up or slowing down of a collection of notes. Grisey eventually fuses traditionally notated music with proportional notation in some parts, while in others he uses long drones (sometimes with extended techniques) with faster passages above them. Although the drones are static in terms of pitch, they induce the sense of motion because of rhythmic changes and the use of flutter-tonguing and other techniques that give a sense of movement.

The composition starts with the attack of the trombone sample and therefore the initial entrance is loud, outlining the low partials of the sound. The higher partials are softer and sustained, eventually fading away. Grisey continues in this fashion with higher partials sustaining and tailing away as lower instruments play the lower partials more loudly. The music slowly coalesces with the focus on the higher partials of the trombone note, with rhythmic figures derived from the original sample. Grisey thus uses the ensemble as a kind of acoustic additive synthesizer, creating a complex spectrum derived from the trombone sample. This composition demonstrates early French spectralism by staying close to aspects of the spectrum and not using material that is not organically found in the sample.

### 1.2.3 Murail's *Gondwana*

Murail's *Gondwana,* composed in 1980, ten years after the birth of spectral music and a year after Hugues Dufourt wrote *Musique Spectrale*, is still considered an iconic example of spectral music. *Gondwana* was written at a time when Additive Synthesis and Frequency Modulation Synthesis had already been created, elaborated and incorporated into software. This allowed Murail to fuse an acoustic trombone sound's harmonics with chords synthesized from a bell tone (inharmonic) using FM synthesis and detailed envelopes. Murail experimented with different carrier frequencies while keeping constant modulation frequencies, creating unique chord clusters and groupings of harmonics/inharmonic pitches (Bennett, p. 16). As in the case of Grisey's *Partiels*, Murail is not trying to recreate the synthesized sound, rather, he uses acoustic instruments to realize a more complex sound originating in an electronic source.

In Murail's notes about *Gondwana*, he shows the equation used to generate different chords, which is the standard C +-(M*I) equation for frequency modulation (Bennett, p. 16). He then shows two series of notes with one as the modulator and the other as the carrier, as well as a list of numbers to be used for the index. On the next page, he shows large chords, including the measure where each chord enters, the approximate envelope, and the scales of pitches derived from the chords (Bennett, p. 17). Murail's process is more focused on the creation of a spectrum through Frequency Modulation than on the analysis of a given spectrum. He also considers the range of the harmonics generated and the instruments that will play them. Brass play the modulation and carrier frequencies while the piano, crotales and other instruments with sharp attacks play higher index notes (Bennett, p. 16-21). Stronger, lower pitches that sustain in the spectrum are played by the brass because they are able to sustain their sound at a constant volume. The higher partials of a sound that fade away are played on instruments that cannot sustain at a constant sound and dynamic.

Unlike Grisey, Murail creates material using an electronic process. However, like Grisey, he uses the ensemble as a giant additive synthesizer to produce complex spectra. Since the instruments create complex harmonic spectra, the recreated spectrum is far more complex than the original. The process of manipulating the spectral information later becomes more prevalent in spectral music and composers combine manipulation and modulation techniques using both electronic and acoustic music practices.[ii]

**1.3 Romanian Spectral Music**

During the period when French Spectralism was becoming established, Romanian composers were experimenting with spectral composition, but in very different ways. Iancu Dumitrescu, and later, his wife Ana-Marie Avram, were at the forefront of what Dumitresco calls Hyper-Spectralism (Clarck, 2009, p. 33). According to Dumitrescu, Romanian spectral music has roots in French spectralism, i.e. Grisey, Murail and pre-spectral composers including Scelsi and Messiaen, with the addition of Romanian folk tradition (Reigle, 2008). Dumitrescu claims that French spectralism relates to serialism because it creates sets of data used as a strict guide to the composition and that French spectralism is more focused on scientific study, i.e. looking at the sound or creating it through a scientific process, then composing with the data. (Reigle, 2008). In contrast to this, Romanian spectralism is not restricted to the starting spectrum, does not always use the spectral elements in transitions, texture or other aspects, and is concerned with the interaction of music with its environment. In addition to the musical properties Dumetrescu describes his phenomenological approach to his work with the following questions, “How does one approach the creative act? How does one music become? What is the validity and authenticity of one’s work? How can we free ourselves from the

---

[ii] For a more detailed analysis of *Gondwana* read Rose (1996).

false? (Noetinger, 1996-1997). Each composition is meant to be an inclusive environment in which electronic sounds, acoustic sounds, and the motions of the conductor and performer are all part of the total experience. Dumitrescu approaches music as an artistic and philosophical experience that can be interpreted differently by each individual listener. Romanian spectral music, in addition to involving pieces for solo instruments and chamber ensembles, also includes a lot of fixed media accompaniment (Noetinger, 1996-1997).

Another contrast between the two concerns notation. French spectralism is traditionally notated, though there is some experimentation with spatial notation, micro-tones and other extended techniques. The Romanian spectral score notations are highly experimental, involving graphic notation and improvisation, and often present a fixed media timeline to guide the performers and conductor. Figure 1.2, for example shows the first page of Iancu Dumitrescu's *Remote Pulsar*.

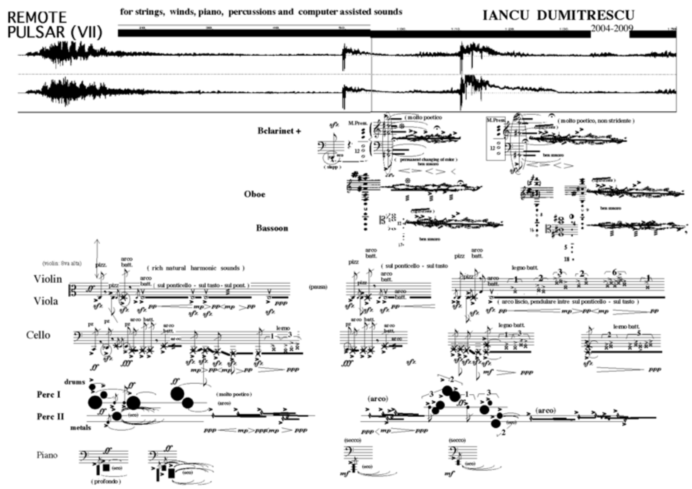



*Figure 1.2 Remote Pulsar by Iancu Dumitrescu (1994 -1996)*

In Figure 1.2 the first two minutes of fixed media are shown on top of the score with the instrumental parts beneath. This fixed media representation serves as a timeline for entrances, releases and other musical gestures. For the instruments, zigzag lines and verbal description indicate tremolo, trill and flutter techniques. Instead of rests, blank space is used to show extended silence. Almost every note has an articulation and dynamic mark, involving a great degree of detail for each note. The percussion is notated on a 3-line staff with varying size circles, diamonds and other shapes, indicating particular instruments, dynamics and expression. It is impossible to determine what sound from the score is part of the spectrum and what is transitional. Even when the sound is contrasting, it does not necessarily mean that the music is not derived from the original spectrum. Dumitrescu seamlessly blurs the line between spectrally-derived material and transitional material leaving the listener to question its origins.

Romanian spectral music composers, such as Dumitrescu and Avram, also compose traditionally-notated spectral scores, yet still use transitional material that is not part of the spectrum. Romanian spectralism allows for the composer to add gestures, ideas or other creative elements not found in the original sound. The use of highly experimental notation also gives the composers the freedom to create their own musical language. While Romanian spectral composers were creating their own processes in the 1970's, their methods and freedom are more closely related to later spectral composers as described in section 2.

**1.4 The German Feedback Group**

The German Feedback group originated in the 1970's as a collective of composers including Johannes Fritsch, Rolf Gelhaar, Clarence Barlow and Mesias Maiguacha. They all had similar musical ideas and created two studios in Cologne: the Feedback Studio and the Feedback Studio Verlag and Publishing House (Mabury, 2006, p. 37). They focused more on melodic ideas than on traditional harmonic ones. They also focused on generating spectra and specific studio style techniques using Ring Modulation and Frequency Modulation techniques using analog equipment (not software like Patchworks), rather than focusing on the analysis of a pre-recorded sound.

According to Anderson (2000), "The influence of Stockhausen is certainly a factor in their music, and the particular varieties of spectral composition found in their work were perhaps triggered by *Mantra* and *Stimmung* (p. 15)". His speculation seems to originate in Stockhausen's use of the overtone series to create harmony and then using the resultant as the basis for melodic shapes (Brandell, 2000). In Stockhausen's *Mantra,* while it is mostly improvisational, Stockhausen focused on changing timbre through analog processing and detailed articulations. However, the pitches in *Mantra* are created through a 12-tone series and its inversion (Sargenti, 2013, p. 567). *Mantra* thereby fuses the idea of timbral manipulation with serial techniques, something traditional spectral composers did not do. While Stockhausen is not considered a spectral composer, he used certain spectral elements, laying the foundation for the Feedback Studios' practices.

The composers from the Feedback Studios experimented with ways of composing melody using sets of generated harmonies, as found in Mesias Maiguashca's *Monodias e Interludios (Anderson, 2000, p. 15).* Here, he generated varying timbres using FM synthesis, then composed monophonic melodies that shift among them (Anderson, 2000, p. 15). The German Feedback composers also used traditional notation and did not follow the experimentalism of the French school nor that of the Romanian school. The other composers of the Feedback studios also employed the fusion of synthesis processes as a means of creating spectra for melodic writing. The Feedback Studios' aesthetics differ from those of the French or Romanian spectralists but, nonetheless, demonstrate an early form of spectral thinking.

## 2.0 Second Generation Spectral Composers and Beyond

Second-generation spectral composers studied with or came after Murail and Grisey. Many went to IRCAM, including Kaija Saariaho, Joshua Fineberg, Steve Lehman and Jonathan Harvey. This generation was thinking about timbre in various ways, and the definition of spectralism started to dissolve leading up to the Istambul 2003 Spectral Music Conference. It was suggested that the term 'spectral music' includes any music that focuses on timbre as an element of musical language or compositional structure (Reigle, 2008).

Damien Poussett says, in *The Works of Kaija Saariaho, Philippe Hurel and Marc-André Dalbavie – Stile Concertato, Stile Concitato, Stile Rappresentativo,* claims that the term 'second generation' is arbitrary and incorrect because the new spectral composers use (watch gerunds!) processes that are very different from those of Murail and Grisey (Poussett, 2000, p.3). The conclusion drawn by Poussett assumes that spectral music was defined by Hugues Dufourt. However, as demonstrated by Murail and Grisey, spectral ideas do not have a set process or characteristic. Poussett's claims are problematic because many second-generation composers use techniques derived from those of Murail and Grisey even though the outcomes are different. Grisey and other early spectral composers did not like the term 'spectral music.' As discussed previously, Grisey wanted the "genre" to be called 'liminal' in order to incorporate all music that deals with timbre. Grisey in this sense foresaw a problem with the categorization of spectralism and wanted to incorporate a broad approach that would include future generations of composers.

**2.1 Kaija Saariaho**

Finnish composer Kaija Saariaho (1952) developed her spectral practices at IRCAM. Prior to finding her voice in spectral music, she had studied at the Sibelius Academy in Helsinki and later with Brian Ferneyhough in Frieburg. While studying with Ferneyhough, she learned about the school of New Complexity and made a very strong statement against it, saying, “All of that complexity, and for what aural result?” (Service, 2012). She also thought New Complexity involved the over-systemization of a compositional process. Instead, she became interested in the work of Grisey and Murail. Saariaho never studied with either, but taught herself spectral music techniques before working at IRCAM in the 80’s (Anderson, 2000, p.20). She went on to develop her own compositional method for creating harmony, micro-tonality and transformation from tones to noise (Burton, 2006). Saariaho can be considered a second generation spectral composer because she uses the spectral practices and inspiration that fall outside of the original practices of Murail and Grisey but are still rooted in their techniques.

One example is her *Oi Kuu* for bass clarinet or bass flute and cello. Here, she explores the commonality between the two instruments’ sounds. Multiphonics notated in the wind instrument meld with the sound of the cello, whose timbres are notated using intricate microtonal notation, bowing pressure notation, moving from tone to noise, doublestops, harmonics, creating airy sounds. Saariaho uses symbols that look like crescendo and decrescendo markings that are filled in to notate bow pressure, while the movement from tone to noise uses an elongated triangle as seen in Figure 1.3.

*Figure 1.1 Left) bow Pressure Notation. Right) Noise to Tone Notation*

The rest of the composition uses standard notation. Saariaho does not use analysis in a traditional spectral way, but plays with the timbres of both instruments in an attempt to find commonality. Often, the bass clarinet/bass flute plays a multiphonic while the cello blends its pitch and timbre to create a sound that is part multiphonic and part cello. Rhythmically, the instruments sometimes follow one another to create a similarity between the parts. Her new way of thinking about timbre became commonplace in spectral music and challenged its definition.

According to Saariaho's biography, she focuses on the slow transformation of sound masses and new ways of harmonic thinking (MacMillan, 2015). Often, she uses recorded sources as the original starting point and composes a sound mass and builds from there. She broke away from Grisey and Murail's ideas by incorporating transitional material similar to those found in Romanian spectral school.

## 2.2 Joshua Fineberg

Composer Joshua Fineberg (1969) has published numerous articles about spectral music and its processes. Unlike Saariaho, Fineberg studied in France with Murail and has a more direct relationship to the original approach to spectral composition. He has also collaborated in a wide variety of fields to study computer-aided composition and perception as they relate to spectral music.

Fineberg uses acoustic models in many of his compositions but says they are more “elusive and sophisticated” than Murail’s or Grisey’s (Fineberg, 2014). In Fineberg’s early work, *Paradigms,* he begins with his own previously composed acoustic instrumental textures and analyzes them. He derives the harmonic material from them and then builds the final composition (Fineberg, 2000d, p. 9). Fineberg is able to take into account acoustic properties that occur during live performances such as natural distortion or reverb when using recorded instruments.

He has composed other pieces that involve computer synthesis and real-time recursive analysis such as *Empreints* (Fineberg, 2000d, p. 71). Here, sound is analyzed to obtain the virtual fundamental and amplitude and the computer then assigns that information to different parameters. The computer accompanies the instrumentalist by playing back sounds that the analysis is triggers (Fineberg, 2000d, p. 72). While this is not purely spectral music, it uses spectralist ideas of analysis and resynthesis as part of a complex musical system. In this example analysis serves as a controller to trigger analyzed sound rather than compositional guide.

Fineberg has also fused traditional spectral techniques, such as frequency analysis with newer technological possibilities, such as machine learning. He has expanded the original notions of spectral music including, for example, improvisation and models systems based on his own pieces combining acoustic research and abstraction. He thus incorporates live electronic processes, computer-aided composition and electronic manipulations that fall outside traditional spectral domains. His use of computer-aided composition also shows a development of a spectral algorithmic process that allows for data mapping and manipulation outside of the traditional spectral techniques.

**2.3 Steve Lehman**

Spectral music, for this second generation of composers, became more of an experimental process than a compositional "attitude" as described by Murail. As a result, spectral music influenced other styles of music beyond those of the western classical tradition. This can be found in the music of Steve Lehman (1978). Lehman studied at Wesleyan University under Alvin Lucier and Anthony Braxton and gained a unique experience in experimental music and jazz. He later received a DMA from Columbia where he studied with Tristan Murail, George Lewis and Fabien Lévy (who had studied with Grisey). It was the influence of this unique mixture of experimental, spectral and jazz composers that enabled Lehman's to incorporate spectral techniques in his experimental jazz. He is also the only Jazz performer who regularly uses spectral processes in his music. These not only give his music a unique sound but shows how versatile spectral processing can be.

Lehman's creates harmonies using spectral techniques as the foundation for improvisation as can be heard on his CD, *Travail, Transformation and Flow*. This is an interesting approach because of the potential strange harmonic progressions that can occur as a result of spectral analysis and its manipulation as well as the microtonality of the harmonics. He uses microtonality, spectral harmonies and sound manipulation as typically found in spectral music side by side with Jazz elements. The music flows the way jazz normally does, but the push towards a cadence or typical jazz structure is not heard.

Lehman uses an octet group in which the drums keep the beat and most of the instruments play block chords in rhythmical patterns that are offset from one another while he "improvises". The ensemble plays certain parts of the spectrum just as in Murail's *Gondwana.* One example of this can be heard in his composition *Echo* where he uses a standard jazz form consisting of an exposition, interlude and coda but repeats each section several times (Lehman, 2012, p.20). Lehman discusses this piece in his paper *Liminality as a Framework for Composition: Rhythmic Thresholds, Spectral Harmonies and Afrological Improvisation* explaining that the spectrum is derived from a vibraphone pitch in which the 15$^{th}$ harmonic is loud and the spectrum is fairly harmonic, with a few instances of inharmonicity (Lehman, 2012, p. 21).

| | |
|---|---|
| Tuba, Bass | $2^{nd}$ Harmonic |
| Tenor Sax | $7^{th}$ Harmonic |
| Alto Sax, Alto Trombone | $9^{th}$ and $10^{th}$ Harmonic |
| Trumpet | $11^{th}$ harmonic |
| Vibraphone | $15^{th}$ and $32^{nd}$ Harmonic |

*Table 1.1 Instrument Division (Lehman, 2012, p.21 - 22).*

Table 1.1 shows how Lehman assigns the instruments to the harmonics of the spectrum. The drums have a modern jazz sound that keeps the rhythm but does not appear to engage the spectrum in any way. Lehman's music does not have any particular spectral sound, but instead sounds like modern jazz. This shows that the influence of spectral music does not create a particular "spectral" sound but offers a way of thinking, reminding one of Murail's description of "attitude."

Lehman does not use the structure of the spectrum to create the form. Instead, as mentioned above, he uses a jazz form consisting of an exposition, interlude and coda and uses beat groupings of 4, 5, 6, 7 and 9 beats, further divided into 4 parts of equal durations (Lehman, 2012, p. 23). This represents a new step in spectral composition as the spectrum is only used to shape the harmonic progression. Lehman's compositional practices can be compared to those of the Romanian spectral composers because of his open use of transitions, and the use of rhythm that originates outside of that suggested by the spectrum.

**2.4 Jonathan Harvey**

Jonathan Harvey (1939 – 2012) did not have a direct relationship with the spectral school. He studied with Milton Babbitt, and during the early stages of his career was dedicated to strict serial techniques (Anderson, 2000, p. 18). During the 1970's and on he became interested in timbre and attempted to fuse serial and spectral techniques.

Anderson (2000) states "… *Inner Light 1* (1971) for ensemble and tape, or *Inner Light III* (1975) for orchestra and tape, he (referring to Harvey) fused the two worlds successfully" (p.19). The electronics in these compositions provide the connection between the spectral world and the serialist world. The tape part moves from timbre to harmony, vowel sounds to harmony and instrumental sounds and transforms one timbre to another (Harvey 1975). The main focus of the electronics is the transformation from one timbre to another, while the instruments reinforce the transformation. Harvey says, "For instance, a trumpet sound leaves the orchestra, changes progressively into a clarinet in mid-flight, so to speak, and returns to the stage area where the orchestral clarinet takes it up" (Harvey, 1975). Harvey is not deriving sets from the sounds but creating sets that go well with the electronics, while transforming the timbres acoustically and electronically. The spectral interest in these compositions is in the transformation of the electronic and instrumental sounds, even though he uses serialist techniques.

Harvey did compose spectral music that is not associated with serialism. *Speaking* for orchestra and electronics is one such piece. Here, Harvey uses custom software called Orchidee, developed by IRCAM, to analyze an audio file of voice sounds and render possible orchestration results (Cont, 2012). Harvey then uses a real-time algorithm in Max/MSP using IRCAM's FTM library to analyze the audio and enhance features related to the voice. A keyboard is used, along with a score follower, to trigger the computer part in time with the orchestra (Cont, 2012). This composition demonstrates a complete spectral and computer-aided compositional process. With the assistance of the computer, the orchestral resynthesis comes closer to the original sound, while the analysis and orchestration possibilities are directly related to the spectrum. *Speakings*, according to IRCAM, involved a two-step process utilizing computer-aided composition and real-time processing. It was part of a larger project called *Making an Orchestra Speak* by Gilbert Nouno, Arshia Cont and Grégoire Carpentier. This involved pure spectral composition because it starts with an analysis of a vocal sound and then attempts to resynthesize it with the orchestra. *Speakings* also demonstrates a "first generation" process similar to that of Grisey's *Partiels* because of the direct relationship between analysis and resynthesis of the vocalizations and the orchestra.

Harvey shows how composers could incorporate past musical approaches, such as serialism, with timbre/spectral thinking. While the music discussed above focused on acoustic approaches, spectral music has also greatly influenced electronic music and other genres ranging from those of pop culture to the avant-garde and more.

## 3.0 Influences on Other Genres and Current Music

With the addition of new software and more powerful computers, spectral processes are both easier to achieve and offer better resolution. At present, while traditional synthesis techniques are still used, spectral manipulation has added many more. Electroacoustic music now uses FFT-based processes to blend or warp timbres. Composers use time stretching techniques, spectral delay, convolution, cross synthesis and many more techniques to alter the frequency and time domain.

An example of intensive spectral manipulation is found in *Out of Breath* by Paul Koonce. Using custom software called PVC, a flute player is sampled breathing and playing one note. Using this sample and custom software called PVC, Koonce manipulated the sample of a flute player breathing and playing one note. He generated all of the other flute notes and then composes what sounds like a flute solo. To do this, the original sound was analyzed and then accurately transposed to all of the desired frequencies. However, the formants of the sound remain in the same place in order to keep the timbre. This is an extreme example of a spectral process.

In pop music culture, spectral processing is used for a variety of effects. Artists regularly alter their voices or instruments with plugins that uses spectral analysis and processing. Antares Auto-Tune, for example, uses spectral pitch shifting to tune a vocal part or create timbral variation. Convolution reverb is often used to add reverb from a particular location into the mix. This is achieved through spectral synthesis that combines the spectrum of the instrument with the reverb spectrum of the desired place.

Spectral processing has become common in many different musical styles because of its accuracy for sound manipulation and variety of interesting and creative effects.

**3.1 Noise and Glitch Music and Its Timbral Uses**

While noise and glitch music are not usually considered spectral music, the process and the artist's approach to sound is similar to that of spectral music in its focus on timbral exploration. Peter Krapp (p. 59) in *Noise Channels,* says "Coining the notion of music concréte in 1948, Schaeffer made looped samples that embrace the sonic givens of modern life world, rejecting the abstractions of musical composition without abandoning musicality" (Krapp, 2011). This offers an early example of an artist focusing on the sound itself. Music concréte could be considered spectral music according to the new definition established at the 2003 Spectral Music conference that includes any music that uses timbre as an element of musical language or compositional structure (Reigle, 2008).

Merzbow offers another example of the focus on timbre as a fundamental aspect of composition. Though his music is largely improvisatory, his focus on the manipulation of timbre over time can be related to Grisey's idea about liminal music. While listening during performances, he uses his ear to bring out salient properties. Merzbow's music does not have pitch or use acoustic instruments. Instead, he uses feedback, audio generators, samples and many types of effects and filters to manipulate the frequency information of the sounds (Christie, 1998). His compositions use transitions that are bound to his equipment's limitations creating a natural compositional boundary that he is unable to cross.

If Grisey had been able to change the name of the genre to liminal music, the genre would encompass noise, glitch, electronic and more. Sound art, such as that of Christian Marclay, who alters vinyl records to change their sound and performs with a turntable, would also fall within this category. This shows that although the original French spectral approach has splintered, it still had a lasting effect that can be traced through compositional process and spectral manipulation techniques.

**Conclusion**

Spectral music began in 1970 with Grisey and Murail seeking to create music that resynthesized its sonic origins in different ways. Originally, Dufourt coined the term spectral music to refer to any music created using spectral analysis. Since Dufourt, the definition of spectral music has expanded to include all music that focuses on timbral manipulation. This new definition is vague, as intended, and focuses on Grisey's 'liminal' music idea. Spectral music is no longer a genre, but has moved towards suggesting a process that defines it, and encompasses music that involves timbre as the main focus. This allows for it to live in many new forms of music. While this original type of spectral music did not endure unchanged, it has led to a field of study called spectromorphology and a new means of music exploration that can be incorporated into any genre of music.

# Chapter II.

Spectral Techniques

## Introduction

Spectral composers use many processes to transform and manipulate spectra. These range from basic pitch transposition to complicated formant analysis and algorithmic manipulations. Composers are building upon these techniques and creating ever more innovative ways of manipulating and producing spectral information to inform their compositional processes.

As mentioned above, Grisey used spectral information in *Partiels*, where an E2 trombone sample was the spectral source (Grisey, 1980). Murail, in *Gondwana,* used frequency modulation to obtain harmonic frequencies and amplitude envelopes (Bennett, p. 7). Grisey's and Murail's compositions are used as examples throughout this chapter because they clearly demonstrate the use of spectral analysis, transposition (*Partiels)* and modulation (*Gondwana)*.

The analysis, manipulation and interpretation of a spectrum form a direct part of the compositional process – with the composer inspecting and shaping the information to create an aural structure. As expressed by many in the spectral community, Peter Krapp (2011) in *Noise Channels* states "Synthesis and composition are the same concept in two different languages" (p. 64). This quote can be adapted to spectral music and turn into analysis and composition are one and the same or inseparable. In spectral music, the analysis forms a crucial part of the process because all of the musical parameters are derived from the spectral information. Therefore, it can be argued that analysis/synthesis and spectral music composition are directly intertwined. This is supported by Murail who referred to spectral music as an attitude rather than a set of techniques (Fineberg, 2000a, 2). This chapter focuses on techniques used to develop and manipulate spectral information in both acoustic and electronic music.

**2.0 Spectral Manipulation**

The manipulation of a spectrum using a mathematical process offers a fundamental technique for altering its content. The resulting spectra, while still related to the original in terms of frequency relationships, duration and amplitude, are altered in their specific frequency, noise elements, density and duration. The following sections (2.1, 2.2, 2.3, 2.4) illustrate a variety of such techniques that use frequency data in Hertz. Section 2.1 shows how manipulating the spectrum yields a similar yet different harmonic structure. In section 2.2, manipulations combine two spectra into a new one that reflects properties of both. Section 2.3 describes how to analyze the spectrum for flatness, formants, centroid and flux. Finally, the construction and reading of a sonogram are explained to lay the foundation for chapter 3 and further compositional processes that rely on sonographic information. In the first and second sections, for the purpose of clarity, the spectra used in all of the examples are analyzed for frequency content before being otherwise transformed.

## 2.1 Transformation and Mapping

Transposition techniques enable the composer to create multiple related spectra from a source spectrum. The processes explained here start with basic techniques that create new spectra with similar harmonic properties. More advanced techniques are then described that expand and contract the source spectrum and render frequency relationships inharmonic or more chaotic. During this section the numbers and techniques used are not for audio processing but the creation of pitch material using frequency in Hertz.

### 2.1.1 Addition and Subtraction

The most basic transposition technique involves addition or subtraction of a frequency. While the frequencies change, the relationships between them do not. In Table 2.1, spectrum A is transposed by 50 Hz to yield spectrum B

| | |
|---|---|
| Spectrum A | 100, 200, 300, 400, 500, 600 |
| Spectrum B Transposition by 50 Hz | 150, 250, 350, 450, 550, 650 |

*Table 2.2. Shifting by Addition*

While Spectrum B is 50 Hz higher than Spectrum A, the difference between frequencies is still 100 Hz. This process thus creates a new spectrum with a higher base frequency but the same relationship between the two.

### 2.1.2 Multiplication and Division

Multiplication and division both transpose the spectrum and change the relationship between frequencies. Multiplication causes a greater distance between harmonics whereas division causes a smaller one. However, because humans hear logarithmically the audible relationship will sound the same. Table 2.2 shows the multiplication and division of a spectrum and the resulting relationships between the frequencies.

| | |
|---|---|
| Spectrum A | 175, 275, 375, 475, 575, 675 |
| Multiplication by 3 | 525, 825, 1125, 1425, 1725, 2025 |
| Division by 3 | 58.3, 91.6, 125, 158.3, 191.6, 225 |

*Table 3.2 Transformation by Multiplication and Division*

Multiplication causes the spectrum to expand by a factor of the multiplication number. In Table 2.2, since the spectrum is multiplied by 3, the harmonics expand by a factor of 3. Using the above example, the difference between the any consecutive frequency is 100 Hz. After multiplication the difference between the any consecutive frequency is 300Hz. Dividing by 3 results in a contraction of consecutive frequencies.

### 2.1.3 Exponents

The spectrum can also be expanded and contracted by the use of exponents. Exponents greater than 1 will expand the spectrum by increasing the distance between consecutive harmonics. This is similar to the case of multiplication, except that the distance between frequencies increases more as the harmonics get higher. Exponents less than 1 will condense the spectrum by decreasing the distance between harmonics as they get higher. In Table 2.3, the harmonics are raised by a power of 1.2 and lowered by a power of 0.7.

| | |
|---|---|
| Spectrum A | 100, 200, 300, 400, 500, 600 |
| Spectrum Raised by the Power of 1.2 | 251.18, 577.08, 938.74, 1325.78, 1732.86, 2156.65 |
| Difference Between Harmonics | 325.9, 361.66, 387.04, 407.08, 423.79 |
| Spectrum Raised by the Power of 0.7 | 25.11, 40.8, 54.19, 66.28, 77.49, 88.04 |
| Difference Between Harmonics | 15.69, 13.39, 12.09, 11.21, 10.55 |

*Table 2.3 Transformation by Exponents*

Table 2.3 shows how the distance between harmonics increases or decreases exponentially based on the exponent's value. An exponent greater than 1 also causes any periodic spectrums frequencies to follow a harmonic series-like pattern by having larger distances in lower harmonics and smaller distances in higher harmonics, although the frequency relationship is not harmonic. Depending upon the frequency content of the original spectrum exponential transformation can greatly warp the spectrum.

### 2.1.4 Other Functions

Any math function can be used to alter a spectrum, but the level of spectral change after the process can increase drastically. For example, a Gaussian Curve can be used to transpose the frequencies by plotting the graph and using the X-axis for the harmonic frequency and the Y-axis (multiply by 100) for the new frequency. Half of the Gaussian Curve should be used in order to prevent the frequency values from doubling.

$$y = \frac{10}{\sqrt{2\pi}\sigma} e^{-\frac{x2}{2\sigma 2}}$$

*Equation 2.1 Gaussian Curve*

The transposed spectrum will have a larger roll-off towards higher harmonics or low harmonics depending on which side of the graph is used. In Table 2.4, a spectrum that moves up by 100 is mapped to 0 – 5 (multiplied by 100) on the X-axis yielding new frequencies by the intersection of the harmonic number and the curve on the Y-axis.

| | |
|---|---|
| Control Spectrum | 100, 200, 300, 400, 500, 600, |
| Gaussian Transposition | 398.94, 242.49, 54.69, 4.43, 0.14, 0 |
| Difference Between Harmonics | 156.45, 187.8, 50.26, 4.29, 0.14 |

*Table 2.4 Transformation by Gaussian Curve*

This altered spectrum may include frequencies that are too low to perceive. On the other hand, these frequencies can be used in electronic music to control low frequency oscillators or modulation frequencies.

As discussed, processes such as adding and subtracting transpose the frequencies while keeping the initial relational structure, while multiplication and division expand and contract it. Exponential transposition breaks the initial structure, causing the harmonics to expand or contract more or less as the harmonic number rises. Any other type of function can be used to shift a spectrum, but, depending on the particulars, this will dramatically alter the relationship of frequencies and harmonics. As shown in the instance of the Gaussian Curve, the harmonics can be mapped up or down depending on which half of the curve is used. The next section discusses the techniques that employ two spectra to produce an entirely new spectrum.

## 2.2 Modulation and Distortion

Modulation techniques in computer music include Ring Modulation, Amplitude Modulation and Frequency Modulation. These techniques yield sidebands as a result of the multiplication of two signals, i.e. carrier (C) and modulator (M), or two signals and an index (I) (Dodge, 1997, p. 90-115). The algorithms that describe the frequencies produced from a modulation process can be applied to spectral manipulation by having two spectra that function as carrier and modulator frequencies.

### 2.2.1 Ring Modulation

Ring modulation is a very common technique in computer music, that is achieved by multiplying a carrier signal by a bipolar modulation signal. Normally this can be expressed as $New\ Sample = f1 * f2$ where f1 is the carrier frequencies sample value and f2 is the modulators sample value. Ring modulation creates two sidebands around the original frequency. This occurs because the carrier is removed while the two new frequencies are created (Randall, 1970 p. 92). The new frequencies are expressed as C+M and C-M. For example, ring modulation using a 263 Hz carrier and 2 Hz modulator produces 265 Hz and 261 Hz frequencies. Frequencies this close are heard as one frequency with a beating pattern. In many cases, when a modulation frequency is above sub-audio frequency, the sound produced involves two separate sounds, because the space between the frequencies is large enough to perceive as two separate entities. For example, if the carrier frequency is 500 and the modulation frequency is 100 the resultant frequencies are 400 and 600. These two frequencies are far enough away to be perceived as separate.

When the mathematical model of ring modulation is applied to two spectra, both spectrum's frequency content is used to create a new spectrum. Since spectra are in Hertz and not samples, there are no bipolar waves. Instead, the C-M and C+M expressions are used to calculate the new spectrum. In order to use this algorithm, the frequency content of the 2 spectra is needed. This can be obtained through peak frequency analysis of an existing sound or made up based on some kind of algorithm, melody, etc. i.e. 261, 440, 783, 932 (C3, A4, G5, Bb5). Table 2.5 shows the results of a ring modulation process on 2 spectra. The first frequency of spectrum 1 gets added to and subtracted from all the frequencies of spectrum 2, then the second frequency from spectrum 1 etc.

| | |
|---|---|
| Spectrum 1 | 100, 200, 300 |
| Spectrum 2 | 60, 120, 180, 240 |
| Resultant Spectrum | 160, 220, 280, 340, 40, **20**, **80**, **140**, 260, 320, 380, 440, 140, 80, **20**, **40**, 360, 420, 480, 540, 240, 180, 120, **60** |

*Table 2.5 Ring Modulation of Two Spectra*

The number of frequencies produced by this method is calculated by the number of frequencies in spectrum 1 multiplied by two, times the number of frequencies in spectrum 2 or (2*S1) * S2 (Fineberg, 2000a, p. 97). This process eliminates the original fundamental and creates a new fundamental at 20Hz in the above example. In some cases, all the harmonics in spectrum 2 may be present in the new spectrum and some frequencies of the new spectrum may be duplicated as shown in Table 2.5. When negative numbers are produced from subtraction, the number can alias, wrap or the absolute value can be taken.

There are different approaches for eliminating or using duplicate frequencies. In ring modulation for sound synthesis, duplicate frequencies usually get louder because their amplitudes add to one another. However, this is not the case in ring modulation for spectral manipulation because it does not take into account the amplitude of the frequencies. Duplicate frequencies can be left out and the remaining frequencies arranged in ascending order or in an order that reflects the sidebands. This will give an overview of the new spectrum and its frequency content. In another approach, duplicate frequencies can have a higher probability of occurring or yield a louder dynamic, as is the case with synthesized ring modulation. Since duplication is possible and highly probable, depending on the spectrum, any number of processes can be used to utilize the duplicate frequencies such as probability of frequency occurring or dynamic emphasis.

### 2.2.2 Amplitude Modulation

The process of amplitude modulation is similar to that of ring modulation but the modulation frequency is a unipolar wave ranging between 0 and 1. Amplitude modulation might be expressed as $New\ Sample = f1 * (sinetone * 0.5 + 0.5)$ where a sine tone is being scaled to a unipolar signal and multiplied by the carrier. The carrier frequency in amplitude modulation remains present in the spectrum as expressed by C, C+M and C-M (Randall, 1970 p. 90). Compared to ring modulation, this process produces more frequencies in the series. The number of frequencies for amplitude modulation is calculated by ((2* S1) * S2) + S1. When applied to the spectra in Table 2.5, the equation will be ((2*3) * 4) + 3. While amplitude modulation creates a complex spectrum, the original frequency information of the carrier spectrum remains part of the new spectrum. Using this technique with one carrier spectrum and multiple different modulator spectra can create many different new spectra that sound similar because of the presence of the carrier spectrum.

**2.2.3 Frequency Modulation**

Frequency modulation was developed by John Chowning in 1970 (Roads, 1995, p. 225). Frequency modulation involves a carrier frequency, modulation frequency and index that are used to synthesize complex spectra. The index affects the spectral richness of the resulting sound and is defined as the deviation of modulation frequency I = D/M. Frequency modulation can be expressed as $y(t) = \sin\big(2\pi f_c + \Delta f_c cos(2\pi t f_m)\big)$. In this equation $f_c$ is the carrier frequency and fm is the modulation frequency (Cook, 2002 p. 117). However, we can describe the frequencies produced as C, C-M, C+M, C+2M, C-2M, etc. or C + (M*I) and C- (M*I) where C is the carrier, M is the modulation frequency and I is any integer (Chowning, 1973, p. 527). Using C-M, C+M, etc. is a simpler way to calculate sidebands when trying to produce spectra from frequency number in Hz. An example of frequency modulation techniques used to create acoustic music, rather than for sound synthesis, can be found in Murail's *Gondwana.* Here, he creates a simulated bell spectrum by using a fixed modulator for six different carrier frequencies. Using the C+(M*I), C-(M*I), etc. to generate harmonies, Murail orchestrates the resultant chords with the addition of a trombone spectrum (Fineberg, 2000a, p. 69).[iii] In the case of spectral music I can be any number.

When using FM techniques for acoustic music, the index becomes the harmonic number starting at 1 (fundamental) and increases by a number to a predetermined maximum. Since the index is being used differently than in FM synthesis, the word ordinal number is going to be used.[iv] Table 2.6 shows the spectrum produced from a 200Hz carrier, 100Hz modulator with ordinal numbers from 1 - 4. When calculating the lower sidebands, negative frequencies can occur as shown in Table 2.6. When synthesizing a sound, negative numbers alias to a positive numbers or are heard as the positive numbers with an inverted phase. To rationalize these negative numbers in Table 2.6, the absolute value can be taken which in turn would make -100 into 100 and -200 into 200.

| | |
|---|---|
| Frequency 1 | 200 |
| Frequency 2 | 100 |
| Resultant Spectrum | 200, 300, 100, 400, 0, 500, **-100**, 600, **-200** |

*Table 2.6 Frequency Modulation of Two Frequencies*

[iii] An illustration of Murail's frequency modulation harmony creation can be seen in *The Principles of Spectralism and Tristan Murail Gondwana* by B. Bennett
[iv] An ordinal number is defined as a number designating the place occupied by an item in an order or sequence (Ordinal Number, 2011).

Sidebands with more complex structures are created by using two spectra just as in other modulation techniques. However, instead of adding and subtracting all the frequencies from spectrum 1 and 2, ordinal numbers are assigned to the frequencies in spectrum 2. For example, in Table 2.7, the first frequency at 50Hz in spectrum 2 will have an ordinal number of 1 and the frequency at 100 Hz will have an ordinal number of 2, etc. This shifts the harmonic structure and forms and irregular pattern.

| | |
|---|---|
| Spectrum 1 | 100, 200 |
| Spectrum 2 | 50, 100, 150 |
| Resultant | 100, 150, **300**, **550**, 50, 200, **450**, **250**, **400**, **650**, 200, **0**, **250**, **350**, **400**, **750**, **250**, **300**, **20** |

*Table 2.7 Frequency Modulation of Two Spectra*

By using C, C-M, C+M, C+2M, C-2M, etc. for each frequency in spectrum 1 and 2, the amount of frequencies in the resultant spectrum grows and a new structure develops. Duplicate frequencies can be created from this process and used in a variety of different ways such as weighted probabilities, melodic figures and synthesis parameters.

### 2.2.4 Enveloping FM Envelopes

In frequency modulation, the index value is often controlled by an envelope. In frequency modulation with two spectra, the ordinal number can be controlled by following some kind of curve. In Table 2.6, the ordinal number is set to the integer number harmonic. However, with the addition of enveloping, subtle harmonic changes can occur over time that follow the shape of the curve as shown in Figure 2.1. Envelope numbers can be scaled to any range for the ordinal number. In FM synthesis, the change in index not only controls the frequency of the sidebands but also the volume of the sidebands over time. Since volume is not a concern in spectral manipulation based in numerical data, the ordinal number value affects the amount of expanding and contracting of the sidebands.

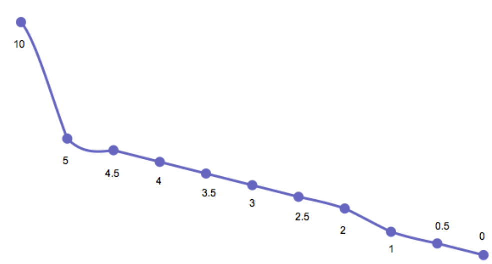



*Figure 2.1 Index Envelope*

In Figure 2.1 an envelope is created and scaled to numbers between 0.0 and 10.0. Each number serves as the ordinal number for a particular harmonic. If the carrier is 100Hz and the modulator is 50Hz the sidebands become 600, 350, 325, 300, 275, 250, 200, 150, 125, 400, 150, 125, 100, 75, 50, 0, -50, -75. Figure 2.2 shows a spectrogram of oscillators playing all the frequencies. This spectrogram is effected by phase that changes the amplitude although all the oscillators are the same volume. The graph on the right shows how the frequencies fluctuate according to the ordinal number values, steadily increasing or decreasing.

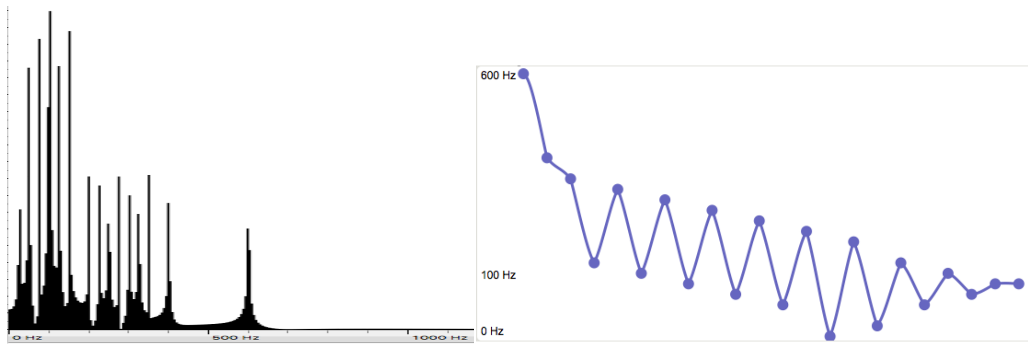



*Figure 2.2 Spectrogram and Graph of Frequency Fluctuation from Envelope Mapping*

Envelopes can also be used for the modulating frequency as well as the carrier. The effects are similar and cause the sidebands to change over time. Just as Murail did, different frequencies can be used for the carrier while keeping the modulator the same or vice versa. This creates spectra with similar pitch relationships but different sideband frequencies and timbre. Changing the modulator over time cause the spectrum to expand, contract or fluctuate depending on the shape of the envelope and the range it is mapped to. The resulting spectrum resulting from frequency modulation range from harmonic to inharmonic spectra depending on the parameters set.

Envelopes can also be created to control volume independently from the creation of sidebands and modulation of frequencies. In the above process, amplitude was not being used, but with the addition of an amplitude envelope the amplitude of the sidebands become controllable.

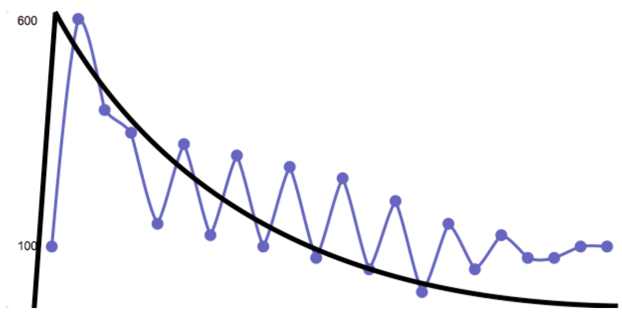



*Figure 2.3 Volume Envelop of Sidebands*

In Figure 2.3 the sidebands created through the FM process are enveloped by the envelope shown in black. This makes the fundamental frequency the loudest, while the rest of the frequencies exponentially decrease in volume. The combination of a changing ordinal number shown in blue and amplitude envelope allows for intricate pitch and dynamic possibilities.

**2.3 Analysis Techniques (Audio Descriptions)**

All modern analyses start with a Fast Fourier Transform that yields an array of real and imaginary numbers. These numbers, passed through different functions, can give information about the sound including magnitude, phase, onset, flux, centroid and others that in turn inform composer choices.

The Fourier Transform, developed by Jean Baptiste Joseph Fourier in 1807 (McClellan, 1998, p. 321), showed that any periodic signal can be expressed as an infinite number of sinusoids whose frequencies are integer multiples of the fundamental frequency and with possibly different phases (Park, 2010. P. 280) or a superposition of complex exponential signals (McClellan, 1998, p. 321). It converts a sound from the time domain to the frequency domain. Fourier's discovery originally applied to continuous signals. After the Fourier Transform was developed, the Discrete Fourier Transform was created to calculate the spectrum of discrete time signals. From the DFT, the Fast Fourier Transform (FFT) was created by James Cooley and John Tukey for greater efficiency in calculating the DFT on a computer (Hardesty, 2009).

$$f_1 = \frac{sample\ rate}{window\ size}$$

*Equation 2.2 Sample Rate of FFT*

Equation 2.2 shows the equation for the center frequency of the first frequency bin in the output of a DFT for a given sample rate and window size. The center frequencies of the other bins are integer multiples of $f_1$. For example, if the sample rate is 44100 samples per second for the incoming audio signal and the FFT window size 2048, then the spectrum of the signal is sampled every 21.53 Hz. This creates bins centered at 21.53, 43.06, 64.59, 86.1 Hz, etc. with a certain magnitude (Klingbeil, 2009, p. 16).

The DFT can also be resynthesized from the phase and amplitude information that was analyzed by using an Inverse Fast Fourier Transform. The IFFT allows for the information of analyzed sound to be manipulated in the frequency domain and converted back into a time domain signal. It is this process that enables time stretching, vocoding, convolution and realization techniques in audio synthesis. However, resynthesis through the IFFT is not pertinent to spectral analysis, therefore the following discussion will focus on real and imaginary numbers and sonograms. IFFT resynthesis will be covered in more detail in chapter 3.

### 2.3.1 Feature Analysis Using Real and Imaginary Numbers

A Fast Fourier Transform yields complex numbers in Cartesian coordinates that are then converted polar coordinates where amplitude is the distance from the origin (0,0) and phase is the angle around the origin (Dudas, Cycling74). Amplitude and phase can be calculated using the real and imaginary parts by $amplitude = (\sqrt{real^2 + imaginary^2})$ and $phase = \arctan\left(\frac{imaginary}{real}\right)$ (Burk, Chapter 3). The resultant amplitude and phase can be manipulated to create a new spectrum or analyzed for different content, such as onset trends and many more.

Spectral shifting, another option, can be used to transpose and manipulate the spectrum. The mathematics of spectral processing is beyond the scope of this dissertation but a basic understanding of the principles will be discussed. Spectral shifting involves analyzing the frequency bins and shifting them on the frequency axis. For example, if a sample rate of 44100 is used with a window of 4096 bins, the FFT bin is represents every 10.76 Hz as shown in Figure 2.5. At each bin, there is an amplitude that reflects the average energy of that bin. In order to shift the spectrum, the amplitudes from the bins can be shifted by a certain factor that will keep the pitch relationship the same but move the spectrum up or down (Burk, Chapter 3). An example of this can be seen in Figure 2.4 where the first 18 bins are shown. This illustrates shifting a spectrum up by a factor of 3.

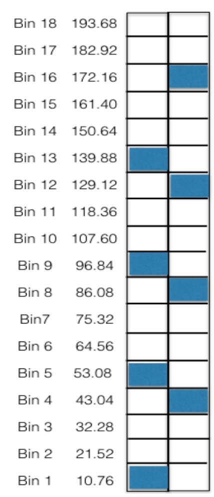



*Figure 2.4 FFT Bin Shifting*

This process of shifting, however, may cause unwanted distortion or "phasiness" if being used in audio because of the deviation of phase between frames. To solve this problem, further processing may be needed to eliminate unwanted timbral distortion.

### 2.3.2 Spectral Flux

Spectral flux is the amount of change in the amplitude spectrum between the current FFT frame and the previous one (Peeters, 2011, 2909). A sustained sound will not have much flux while a percussive one will have a lot. This analysis serves as a guide to understanding the change in the sound over time and when a new sound occurs or passes a certain threshold.

To calculate the spectral flux, the previous real and imaginary numbers in an FFT frame are subtracted from the current FFT frame and then the square root is taken: $flux = \sqrt{curentFFTWindow - previousFFTWindow}$ (Lechner, 2014, p. 253). The formal equation is $flux = 1 - \frac{\sum_{k=1}^{K} a_k(t_m - 1) a_k(t_m)}{\sqrt{\sum_{k=1}^{K} a_k(t_m - 1)^2} \sqrt{\sum_{k=1}^{K} a_k(t_m)^2}}$ where $a_k(t_m)$ is the $k^{th}$ bin of the amplitude for the $m^{th}$ window (Peeters, 2011, p. 2009). Frames that change a lot will result in a higher number while frames that are the same will cancel out. Spectral flux shows the difference between frames and the square root in the above pseudo-equation serves as a scaling mechanism to simulate logarithmic values.

This short algorithm creates numbers between 0.0 and 1.0 that reflect spectral change. Numbers closer to 0 show that the spectrum is steady while numbers closer to 1 show an abrupt change in the sound. The onset of a sound will be detected as a number close to 1 while the middle of a sine tone will be 0 because every frame in a sustained sine tone is the same, so when subtracted from one another the result is 0. Spectral flux is used in software such as Mixx or Virtual Dj to sync music or match beats within a live set. It uses onset detection to calculate where the beat is so the songs/sounds can be aligned and melded together.

### 2.3.3 Spectral Centroid

Spectral Centroid calculates the center of mass of the amplitude spectrum. This is done by calculating the weighted mean of the frequencies in the sound (Peeters, 2004, p. 11). It also reflects brightness because high frequencies with large masses create a sound with a bright timbre (Grey, 1978, 1493). For musical signals the spectral centroid is much higher than the fundamental frequency (Nam, 2001).

The algorithm for spectral centroid takes the average frequency and divides it by the sum of the amplitude (Nam, 2001, 1). In order to calculate the centroid of a spectrum, $centroid = \frac{\sum_{k=1}^{K} k\, a_k(t_m)}{\sum_{k=1}^{K} a_k(t_m)}$ is used (Peeters, 2004, 11). This can be done by summing the magnitude of the FFT and dividing it by the sum of the FFT at a particular bins.

### 2.3.3 Spectral Flatness

Spectral Flatness is a feature that shows how close to noise the sound is. To calculate flatness, the geometric mean is divided by the arithmetic mean: $flatness = \frac{(\prod_k a(k))^{1/k}}{\frac{1}{k}\sum_k a(k)}$ (Peeters, 2004, 20). The resultant measurement is a number between 0 - 1 with 0 being no noise and 1 being noise. The number created through this analysis can be used further to measure tonality by $flatness\ dB = 10 * Log_{10}(flatness), \min\{\left(\frac{flatnessdB}{-60}\right), 1\}$. This will give numbers between 0 -1 with 1 being tonal and 0 being noise (Peeters, 2004, 20). If the sound being input is a sine tone the number calculated for flatness will be 0 because it is a tone while white noise will be close to 1. For the tonality measurement the numbers would be flipped. This calculation is done for all of the bins in the window and then starts calculating again with the new window

The measurement taken does not have a direct relation to a controllable parameter in electronic music or acoustic music. Instead, the number can be used as a noise filter for any part of the spectrum that is noisier than a set threshold. The numbers can control the volume of the sound making the sound louder with more/less noise. Spectral noise can also inform the composer about timbral changes in the spectrum by allowing the tracking of the increase or decrease of the calculated number.

### 2.3.4 Formants and Peaks

Formants are concentrated amounts of energy around different frequency bands caused by the acoustic sound's natural resonance (Wood, 2005). Formants are a major contributor to timbre, enabling listeners to perceive the source instrument while hearing pitches in different ranges (Fineberg, 2000a, p. 88). On a sonogram, formants are represented with a more intense hue in relation to the surrounding one. The formants of a sound also show a center of mass in the spectrum that may or may not be related to the centroid. One major difference between formants and peaks is that peak frequencies may not be formants because they do not have concentrated spectral energy. For example, a sawtooth wave analyzed with a peak detector will show the harmonics of the wave while a formant analysis will show no information because there is no concentrated spectral energy.

Analyzing peaks and formants requires an FFT to obtain the amplitude of different bins and then filter out everything below a certain threshold. The bin can then be analyzed for frequency content. In Table 2.8, the frequencies and amplitudes are given in two columns and the formants are highlighted. It is easy to identify the formants because the amplitude is much higher than that of the surrounding frequencies. The sound used in Table 2.8 is a Clarinet pitch B3. In the sample, the sound has a concrete pitch and some air noise. As seen in Table 2.8, the fundamental pitch is 250.783 Hz (B3 ~4 cents sharp) with an amplitude analyzed at 4.084 dB. The next loudest frequency is 503.112 Hz, approximately 1 octave higher than the fundamental with the surrounding frequencies and amplitudes at low volume levels. The other frequencies in the table are then from other sources such as the resonance of the instrument and small pressure differences from the performers air and reed. These frequencies are not audible and add more to the timbre than clarinet pitch. Another formant between 1500 Hz and 2700 (not shown) is also a major timbre defining formant for the clarinet (Holz, 2013, p. 11). The 1500 Hz formant occurs because of the resonant frequencies of the clarinet vibrating with the pitch being played. All acoustic sounds produced from a vibrating body will have formants that are created by the resonant frequencies of the body (Woods, 2005). The properties of the individual instrument, *i.e.* the material it is made from and production of sound, will change where the natural resonant frequencies are and affect where the formants are in the sound. Other factors such as humidity can also affect exactly where the formants are. This is because materials expand or contract with fluctuations in humidity (Holz, 2013, p. 10). This is why all resonating bodies sound different from one another and also why instruments in the same family sound similar.

| **Frequency** | **Amplitude** |
| --- | --- |
| 143.272 | -2.971 |
| 250.783 | 4.084 |
| 325.856 | -2.124 |
| 351.081 | -2.862 |
| 372.544 | -3.746 |
| 411.222 | -3.637 |
| 428.591 | -4.006 |
| 460.616 | -4.416 |
| 503.112 | -0.696 |

*Table 2.8 Formants of a Noisy Spectrum*

All of these elements can be calculated and used for compositional purposes. The information obtained through a FFT leads to additional processes for analyzing onset, loudness, and more. These numbers hold the key to complex spectral processing and to the creation of new timbres as discussed in Chapter 3. In the next section visualization of the spectrum will be the focus as will a discussion of how the FFT is used.

## 2.4 The Sonogram

A Sonogram is a graph that serves as an important tool for understanding and gathering a sound's frequency and amplitude information over time. The first visual representation of sound was created in 1857 with the invention of the phonautograph by Leon Scott (Ungvary, 1992). While the phonautograph did not give spectral information, it was the first visualization of sound waves. By 1890, Hugo Pipping was able to obtain spectral information such as amplitude and phase from recordings of vowel sounds (Ungvary, 1992).

It wasn't until 1967, that Fritz Winckel created the acoustic sonograph. The first analogue sonogram was a bank of bandpass filters in which the amplitude of the individual filters controlled a transducer that would sketch onto a piece of paper (Truax, 1978). This process is similar to that of a heterodyne filter with a bank of narrow bandpass filters, often used in radio. The user then selects a base frequency to allow the fundamental and all of its harmonics to pass through the filter bank (Long, 2016). This first analog sonogram had a very low resolution because the maximum number of filters is finite, and thus cannot represent all frequencies. Once digital FFT analysis was created it greatly enhanced the resolution of the sonogram and allowed for a clearer understanding of the structure of the sound. The digital sonogram is not limited by a physical filter bank, but by processing power. It can also be used on a sound with an unknown fundamental or on a sound that is evolving. This also allows more precise representation of noise and other non-harmonic properties within the spectrum.

A digital sonogram uses an FFT to obtain the amplitude and phase information that can be graphed at different indices (bins). The amplitude value of each bin in the FFT is graphed with varying degrees of brightness to a particular bin index representing a small range of frequencies (equation2.2) while the phase is not reflected in the sonogram. The index is virtually the bin number on the Y axis which is the reason the sonogram appears to show frequency. To make understanding the sonogram easier, a frequency scale is usually added to the side. This can be done in Max/MSP by capturing the amplitude and bin number through an FFT and writing the information to a jitter matrix (Charles, 2009). This Max/MSP implementation offers both real-time and non-real-time representations as well as a brightness control that helps identify subtle changes.

A sonogram without a frequency point of reference is still useful in understanding what the spectrum looks like and how the overall frequencies interact in time. Figure 2.5 shows a sonogram of a square wave at 440 Hz.


*Figure 2.2 Sonogram of Square Wave*

The sonogram on the left shows the 440 Hz at the bottom as the thickest, darkest line. This shows the loudest frequency and also that the harmonics are evenly spaced throughout the entire spectrum. The sonogram on the right shows another square wave with a little filtering and has a frequency scale on the left. Here, we see that if the wave is 440 Hz then the 10$^{th}$ harmonic (around 5000 Hz) and up are relatively softer and the frequency scale gives us an estimate of what frequencies those are. Since this is a square wave it is easy to figure out the frequency (440 * harmonic number), but for inharmonic waves and complex waves, the information can only be estimated.

Many digital sonograms give a frequency readout where the cursor is located on the graph or use a peak tracker to obtain the information for the loudest frequencies. This allows for concrete pitch information to be obtained, eliminating the need to estimate information from a complex timbre. The color contrast on a sonogram represents the amplitude level of frequencies. Sonograms allow the composer to see all of the frequency and amplitude information at quick glance and select interesting parts for further analysis and manipulation. Spectral centroid, spectral flux, loudness and other descriptors can also be added as readouts to give the composer a concise illustration of the spectral properties. Sonograms allowed Grisey and early spectral composers to create a unique music compositional practice rooted in the science of acoustics.

**Conclusion**

Spectral analysis and manipulation give the composer tools to understand a sound source and a process for manipulating it. Spectra may be altered in ways that expand, contract, transpose or otherwise elaborate on the existing spectral relationships. Spectra can also be created through frequency modulation techniques and then further manipulated. As mentioned above, Murail created a bell spectrum for Gondwana using a frequency modulation process and did not alter the resulting spectrum in any way. However, in *Partiels,* Grisey analyzed a trombone pitch spectrum and used different transposition techniques to elaborate and expand upon it. These techniques laid the foundation for spectral manipulation and analysis techniques in both acoustic and electronic music.

# Chapter III.

## Image Translation to Spectral Information and Sound Generation

### Introduction

As mentioned in chapter 2, a sonogram is an image that shows the frequency and amplitude of a sound and how it changes over time. There is virtually no difference between a sonogram and a digital image because both can be viewed as pixels with width x height dimensions. Images do not have time values and are normally observed as entities. For example, when one looks at a Jackson Pollock painting, one typically views the painting as a whole and then observes details such as colors, lines and depth. Typically, a viewer does not start at one side of the painting and scan to the other side in an effort to view important information. Sonograms, however, are interpreted from left to right. What if we were to apply the same analytic technique to an image as we do to a sonogram? What kind of intricate and complex sonic patterns emerge?

Peter Krapp (2011) writes, regarding Ryoji Ikeda and Carsten Nicolai finding an oscilloscope that produces shapes when playing glitch sounds, “This accidental discovery pointed Ikeda and Nicolai to the concurrent practice of visualizing computer glitches and to transcoding practices that sonify images and depict the proportions of the sound file” (p.61). This practice shows a shift of focus from purely sonic ideas to ones that take root in or focus on images and information transcoding. Ikeda and Nicolai designed sound in response to how the sound looked when graphed rather than creating sounds from the images.

Aphex Twin's piece $\Delta M_i^{-1} = -\alpha\Sigma_{n=1}{}^{N}D_i[n][\Sigma_{j\in \mathrm{C}[i]}F_{ji}[n-1] + F\mathrm{ext}_i[n^{-1}]]$ released in 1999, has a small section during the last 30 seconds in which a face shows up in the sonogram. In writings and forums about this section of the piece, it is referred to as the "demon face" (Niinisalo, 2010). While listening to the section, it is hard for the listener to visualize the "demon face". Instead, it results in an interruption of the music before decaying away. It was quite a shock when people started to look at sonograms of Aphex Twin and found this. In Venetian Snares's *Look* released in 2001 different images of the artist's cat lined up side by side provided the sound source for the music. Prior to Aphex Twin drawing a face in the spectrum or Venetian Snare's using his cat, composer Iannis Xenakis programmed the UPIC system in 1977 to take drawings and map the lines to different pitches, timbres and samples (Roads, 1995, p. 331). These composers and artists did the opposite of what Ikeda and Nicolai did by using the images as sound generators rather than designing a sound and seeing how it looked. Xenakis used UPIC to experiment with a new compositional practice while Aphex Twin did it to be unique and to add an "Easter Egg" his music, while Venetian Snares's *Look* is more of a novel piece about his cat.

As technology progressed and digital synthesis software became commonplace and more powerful, programs such as Metasynth, PhotoSounder, High C and Iannix allowed artists like Aphex Twin and Venetian Snares to use images to generate sound. Metasynth and PhotoSounder read an image as a sonogram, or give the user the ability to draw a sonogram and produce an electronic sound. When the sound is played back into a sonogram the image reemerges as glissandi and noise, outlining different aspects of the picture. Iannix traces an image allowing the user to receive OSC messages of the traced information that can be mapped to any musical parameter while High C is a modern rendition of UPIC.

The focus of this chapter is the use of image as sonogram and sound generator for music composition. This differs from the practice of Aphex Twin or Venetian Snares because the sonogram is being used to create material for acoustic music composition and to build sound generators created by interpreting pixel information within a usable sample range for electronic composition. The first section illustrates how to use an image as a sonogram and extract spectral features for acoustic music composition. The second section illustrates how to read an image as pixel data and use it as a sound generator for electronic music. This chapter is accompanied by Appendix V that shows a suite of software I developed that can read, draw, trace and erase an image as well as analyze the sound produced. All of the software was developed from my Image to Sonogram software and shows how the framework is adaptable to different visual and sonic ideas.

### 3.1 Image to Sonogram

The focus of my research involves the conversion of digital images into sonograms and, from there, into sound. To accomplish this, I developed a suite of Max/MSP patches called Image Spectra. These programs allow for an image to be turned into sound, traced for salient features or erased and the resultant sound analyzed for spectral centroid, spectral flux, loudness, noise and peak frequency. The interpreted image produces complex sound and rhythmic patterns because of the change in foreground/background, color and focus. The parts of the image that have more brightness are higher in amplitude while dark colors are quieter. The amount of blur in the image will also turn frequencies into pseudo-noise because the blur will obscure any defining characteristic or contour. Any part of the picture in the upper half yields frequencies from 10,000 Hz to 22,500 Hz. Though many of the frequencies are too high to hear, they will affect the overall timbre and analysis. The sounds created from the images are spectrally lopsided when compared to acoustic or most electronic sounds because the high frequencies will not tail off and there are generally many more frequencies in the high range than in acoustic sources. The high frequencies are not related to a harmonic series or a harmonic/inharmonic pattern so they are unpredictable.

Interpreting an image as a sound requires the conversion of image data to amplitude and time data. To do so, the image has to be converted into a 2-plane matrix of information representing amplitude and phase. The dimensions of the image can be expressed as dim $= Amount\ of\ Frames\ x\ FFT\ Size$. This data will then be read frame by frame linearly from left to right and resynthesized with the Y axis as bin index, phase and amplitude information. Just as in the case of regular FFT processes, the overlap and number of frames affects the smoothness of the sound and, in the above case, the resolution of the image. Having a small overlap allows for a smoother transition between frames if the sound is being played back. However, too much overlap will cause the image to become skewed. A small number of frames causes the image to become choppy and unrecognizable, while a high number of frames make the image smooth. The notable change from one frame to another with a small number of frames causes more artifacts in the sound.

The FFT size controls frequency resolution with a higher number dividing the image vertically into more frequency points. Normally, when working with FFT data, the higher the FFT size the lower the rhythmic resolution. However, because the data is generated from an image and playback speed can be controlled independently, the rhythmic accuracy is not important. Since a lot of frames are needed to cover the entire image accurately, the frames tend to be small and rhythm can be varied by adjusting playback speed. Frames can also be paused or selected, creating a more abstract compilation of the image. Having a higher FFT number thus allows for a more accurate image translation into frequency/time information.

### 3.1.2 Image Matrix

Every image matrix has four values: alpha, red, green and blue ranging from 0 - 255 to control each pixel, while the alpha channel is between 0 - 1. Digitally, these values can be assigned to different planes within a matrix, i.e. 4 colors (ARGB) to 4 planes, 2 planes with AR and GB being stored together, 2 planes with alpha in one plane and everything else in the other or any permutation. These planes can be individually manipulated for different image effects such as blur, color change, and filtering. Each planes information can also be stored in a buffer and manipulated independently from one another.

To generate sound from an image, an FFT is used to take 2 different matrix planes that serve as the phase and amplitude information for resynthesis. The index from the FFT is used to gather the amplitude and phase information at the individual FFT indices for resynthesis. In Figure 3.1 a 2 plane matrix is used that has the dimensions of 14x512 to illustrate how an image is broken down into frames and bins with low resolution.

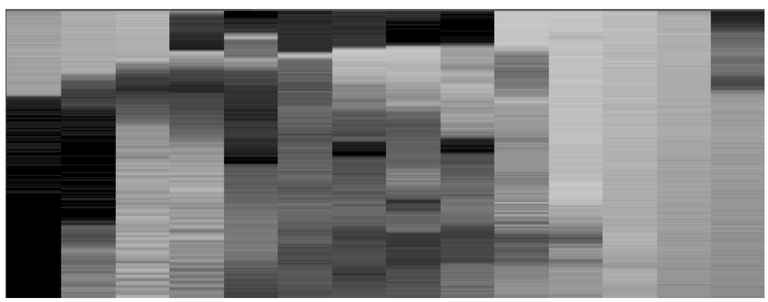


*Figure 3.1 Image Grid*

The FFT size of this image is 512 (512 points on the Y- axis) while there are 14 frames (X-axis). Each frame is an FFT window with no overlap as shown in figure 3.1. In this example there is no obvious emerging image because of the low number of frames and what is shown only hints at what the image may be. Being able to see the individual frames indicates that there is too much information to fit within the particular resolution. Within each frame there are also fine lines that are the individual bins on the Y- axis. The thick black portions within the image will have an amplitude value of 0 while the grey values will range from 0.0 < grey < 1.0.

In Figure 3.2 the same image with 1000 frames and an FFT size of 4096 is used. The individual bin lines get lost because there are a lot more indices that smooth the image on the Y-axis. This allows for the details in the image on the Y-axis to be clear enough that an image emerges with minimal distortion. There are also 1000 frames on the X-axis giving the picture a smooth transition between frames and avoids a choppy sound. The more frames used the smoother the playback is because the changes in the images are represented in different frames instead of interpolated into one.

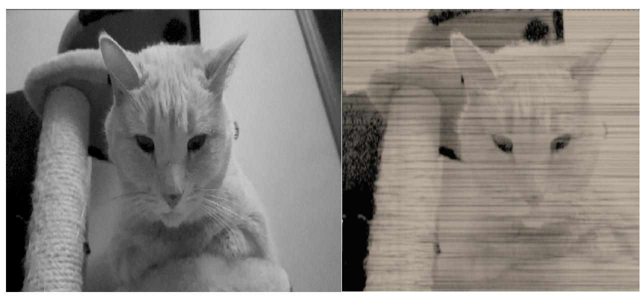


*Figure 3.2 Higher Resolution Image and Resynthesized Sonogram*

This image, when resynthesized and analyzed with a sonogram, will look like the original image due to the high X and Y resolutions. The resolution of the image also reflects the accuracy of the sound. If the resolution is set an appropriate size, then when the sound is captured and reanalyzed in a sonogram it should look like the original image in sonogram form.

### 3.1.3 Turning an Image into a Sonogram in Max/MSP

To build a system that reads an image and plays a sound, two parts have to be built; the first reads the image, the second resynthesizes it. As mentioned above, the first part is built by loading an image into a 2-plane matrix with variable dimension sizes. The frame size defaults to 200 to give the user a low resolution that can be further adjusted. By starting with a low resolution, computers that may not have sufficient power can still run without crashing or lagging. The values of the alpha and the rest of planes are separated from the 4 plane matrix the and the planes 2 - 4 can be used (any singular plane besides alpha can also be used with minimal differences in sound). Alpha will show up as a solid color with no image in it. This separation will create a black and white image with varying shades of grey representing all the colors. Information is not lost by doing this and it is a similar process to conversion of an image into black and white.

The separated information results in an upside-down image that needs to be inverted to read properly. The inverted image then has to be re-entered into a 2-plane matrix which serves as a guide to the resolution output. In the new image matrix, the frame dimension (X-axis) and FFT size (Y-axis) are now controllable and can be altered to manipulate the image. Before the output matrix, any number of effects can be programmed to manipulate the image. These include XY blurring, crossfading, compositing and many more. The output matrix takes the information and packs it into 2-planes that are used as the new image information for resynthesis. In Figure 3.3 the image to matrix Max/MSP programming is shown on the left and the same program with an X,Y blur effect on the right.

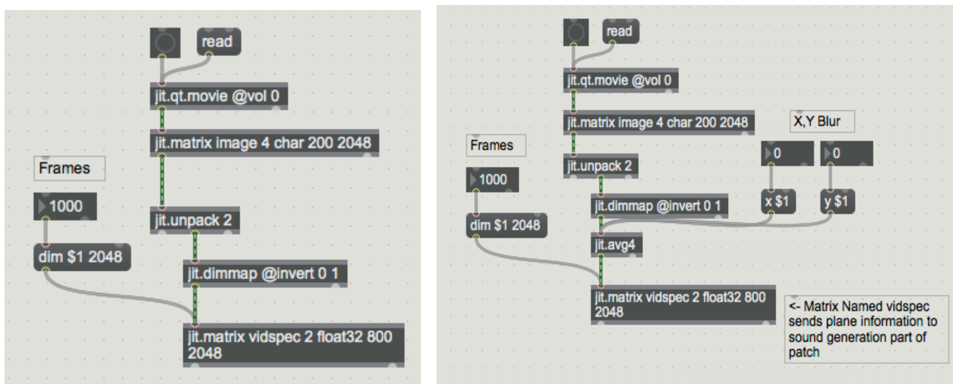



*Figure 3.3 Image to Amplitude and Phase Information. Left) Image to Usable Matrix. Right) Addition of a Blurring Effect.*

For further sound manipulation, the jit.dimmap can be used to turn the image upside or backwards, inverting all of the frequencies. The last jit.matrix object named vidspec enables the software in the sound generation side of the patch to collect the information of the last matrix for resynthesis. Plane 0 will be used for amplitude values while plane 1 will be for phase.

### 3.1.4 Generating a Sound from an Image Based Sonogram

The sound generation part of the patch uses a sample and hold algorithm to control the index being read (Figure 3.4).

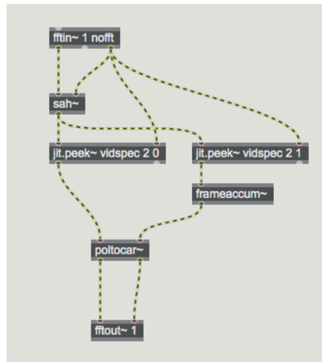



*Figure 3.4 Sample and Hold Control*

Every time the index changes, a value is passed through the sample and hold and triggers jit.peek~ to send a value for plane 0 and plane 1. The values of plane 0 correspond to amplitude while plane 1 is phase. This value is then converted from polar coordinates to Cartesian coordinates for resynthesis. The phase works the same way. However, because phase changes over time, a frame accumulator is used to calculate the running phase. In Figure 3.4 the planes are read in the jit.peek~ object and converted back into Cartesian coordinates before resynthesized with an IFFT (fftout~). jit.peek reads a matrix the same way an audio signal would be read from a buffer, allowing for the index and value to be calculated. The method used to read the matrix values is the same method as that in wavetable playback, time-stretching, phase vocoding, cross synthesis, etc., but differs because the information is in matrix form, not a buffer.

In order to play the sound back linearly, a counter is used that counts from 0.0-1.0 and gets multiplied by the number of frames. Adding the multiplication allows for resolution scaling of the audio and proper time playback. By having a counter, the multiplied number indicates which frame is being played and therefore can be paused or played back at various speeds. To check the system, the image being played back should resemble the initial image when graphed.

While the sound is playing from the image, spectral centroid, flux, loudness, noise, etc., can be analyzed to obtain the characteristics of the resynthesized sound. Analyzing the created sound is interesting because the rules of acoustics do not apply. The spectral structure of the sound embodies the features of the image. These can be rapidly changing or sustaining depending on the scene. The spectra produced using this image-to-sound method creates noisy, glitchy sounds that are quasi-periodic and have many high frequencies not normally present in acoustic-derived sounds. Frequencies also decay and sustain independently from one another, making the structure of the spectrum very complex. In Figure 3.6 an image is run through the process and graphed back into a sonogram.

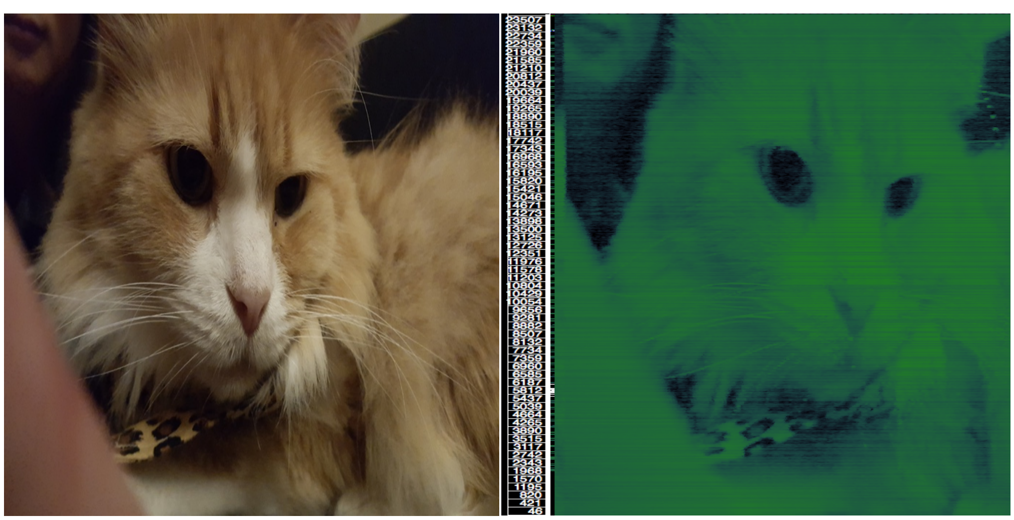


*Figure 3.6 Image to Sound to Sonogram*

In the sonogram, the colors range from black to different shades of green because most of the color in the image is a constant warm orange and there is no light accenting any colors or features. The white on the cat's face is shown as more yellow in the sonogram but is subtle because the difference between the orange and white is small. Gain can also be adjusted and will attenuate all frequencies and make peak frequencies more apparent. Further, filtering can be used to bring out the peak frequencies or add more contrast between color. By filtering out different colors or frequencies, the analysis can be more accurate in capturing a certain part of the image.

### 3.1.5 Other Software

By altering the framework, I created multiple varieties of image to sound software. By keeping the sound generation component of the framework, I created additional software that allows the user to draw or erase a picture. The last component of my software analyzes the resultant the sound for spectral centroid, flux, volume and flatness as described in chapter 2. The drawing and erasing use the same principle as the Image to Sonogram software but add different capabilities before the matrices are parsed. Drawing an image uses jit.lcd in Max/MSP to draw lines on a blank matrix and then sends the matrix information into the vidspec matrix as described above. In the case of the erase software, the color white is loaded so all FFT bins will have maximum amplitude. jit.lcd is then used to draw black lines into this matrix, erasing particular bins. How I created this software in Max/MSP is illustrated in Appendix V.

**3.1.6 Other Uses for Image to Sound Conversion**

Converting image or video information into sound has applications beyond the arts. For example, I have been involved in a year-long chemotactic study in which I altered the Image to Sonogram technique to accommodate video. Chemotaxis refers to the movement of a cell or organism in response to a chemical stimulus (Parkinson Lab, 2003). Dr. Roseanne Ford of the University of Virginia directed this research. I wrote the software to track *E. coli* bacteria's movement in two conditions: as a control and with the addition of chemical attractant. Undergraduate Justin Peruzzi conducted the experimental work with support from a Harrison Undergraduate Research award.

In theory, when an attractant is introduced into the medium, the bacterium's swimming characteristics change; on average, the duration of the run times increase. Under the control condition the equation for the motility coefficient $\mu 0 = \frac{v^2 <r>}{(1 - <\cos(\theta)>)}$ describes the random walk of a bacterial cell, where ν is velocity, <r> is the mean runtime, and $\cos(\theta)$ is the average value of the cosine of the turn angle (Lovely and Dahlquist, 1975). The motion of the bacterium traces out a random walk pattern in 3 dimensions as shown in Figure 3.7. In the presence of a chemical stimulus the random walk is biased in a particular direction eventually leading to or away from the chemical introduced.

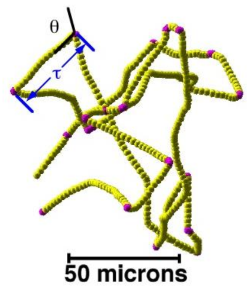



*Figure 3.7 Swimming Path Traced by an E. coli bacterium. From Frymier (1995).*

In this study, when observing a suspension of bacteria there are so many bacteria moving simultaneously that any pattern is indiscernible to the eye, but mapping the information to sound makes subtle differences invisible to the eye more perceptible to the ear.

The same Max/MSP framework as described above is used to capture the video and turn it into a sonogram. However, certain changes were made to accommodate video playback. First, each video frame was divided into 1000 FFT frames and scanned at 26 video frames per second. Second, all video manipulation effects or image processing effects were removed except background filtering. This filtered out any reflection from the microscope and noise caused by out-of-focus bacteria. Lastly, spectral flux and spectral noise analysis were added for real-time analysis. By scanning at 26 times per second, the whole image is seen as a little sliver in the overall audio sonogram. This allows for the sonogram to represent the entire video in one sonographic image. Every second represents 26 frames of the image and the sound in the spectrum shows where the bacteria are in terms of X and Y.

The sound once captured is plotted back into a sonogram to see how global trends change. The control sound looks like random dots scattered densely across the entire spectrum as seen in Figure 3.8 (left side). With the addition of the attractant, there are large holes in the spectrum and the general density is a lot less (Fig 3.8, right side). This is because when the attractant is introduced, the bacteria tumble less frequently, making them less likely to be observed in the focal plane of the microscope. When the video is filtered anything in the background and foreground is filtered out. This makes the control dense because the tumble motion in 3-dimensions is in focus more than it is out of focus. By listening and looking for changes in density, global trends emerge as the bacteria react to the attractant. In Figure 3.8 the control is shown on the left. It has a high density because the bacteria tumble more frequently and more bacteria are observed in the focal plane at any given time. On the right the bacteria are responding to the chemical attractant which causes them to tumble less frequently. They are out of focus for longer periods of time causing less density.

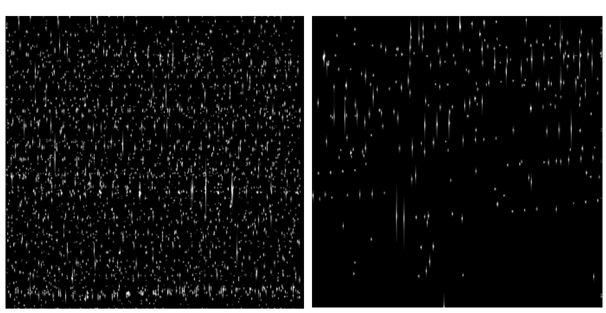


*Figure 3.8 Left) Control with High Density. Right) Response with Less Density*

When the spectrum is analyzed and plotted back in scatter plots, the spectral noise is very high in the control sound and very low in the resultant sound. This is because the random motion of the bacteria causes a lot more noise to appear in the spectrum because of the number of frequencies playing. The attractant sound has less noise because the change in the frequency information does not happen as frequently. The spectral flux for the control sound is high because the number of frequencies is changing very quickly. In the resultant sound the holes in the spectrum cause less significant change to occur between frames. Occasionally the bacteria on the slide will line up in such a way that the spectral flux is large even though the surrounding flux is low. Generally, there are not many peaks for the attractant sound so one peak shows that there was a large change in density from one frame to the next.

These results prove that it is possible to track global trends of bacteria using sound and that the bacteria are reacting to the attractant in an observable way. The results do not indicate which direction the *E. coli* population is moving in, so further analysis is needed.

**3.2 Image to Buffer**

Image matrices consist of X * Y number of points that can be read in a variety of ways. In the previous section, I showed how the image is transformed through an FFT to amplitude and phase components and played back. In this section, the pixels are read as a string of numbers that serve to interpolate a buffer using the Supercollider programming language. Using a buffer to store the image information is more computationally efficient because it doesn't involve the Fourier Transform, but the sound does not reflect the image back when plotted as a sonogram. The resulting sound is also a harsher noise-based sound with unpredictable changes in timbre.

All images have ARGB values and each value for RGB is a number between 0 – 255 and A between 0.0 – 1.0. In 32-bit float encoding, each color channel has 32 bits and 128 bits per pixel that can be negative or positive values (Pixel Formats, Derivative). These values are read from the top left hand corner to the bottom right hand corner of the image. The individual pixel values are then assembled as a long string of numbers (the length is X * Y). The values are then scaled to usable sample numbers between -1.0<n>1.0 otherwise the sound is so "hot" that the system breaks. In Figure 3.9 the code illustrates how a buffer with numbers outside of -1.0<n>1.0 can be used with a limiter to prevent the system from failing.

```
SynthDef("image",{ arg out=0,bufnum = 3, level = 0.1,
dur = 0.01, freq = 800;

var i, limit, t, pan, verb;

i = PlayBuf.ar(1, bufnum, freq, loop: 0);
limit = Limiter.ar(i, level, dur);
Out.ar(out, limit);

}).send(s);
```

*Figure 3.9 Using a Buffer With Numbers Outside a Traditional Range*

Each image has different maximum and minimum values because of the different colors, so the numbers have to be analyzed in order to interpolate the buffer. Proper interpolation is the most important part of the process because it determines aliasing and distortion in the timbre and can greatly affect the stability of the system when not using a limiter. If the buffer is created with any values above 1 or below -1, the sound can clip or, depending on the size of the number, send a very loud signal that can cause hearing or hardware damage. If the buffer is interpolated without scaling the values or scaling them between –x and x, a limiter can be used as in Figure 3.9 to bring the signal to an appropriate level. By doing this, the waveform changes or is wrapped/clipped to produce different sounds. If the waveform starts with a sample that is too large the sound will repeatedly clip when looped back. By overdriving the buffer and using a scaling method to bring the signal back down, very interesting, raw, distorted sounds can be created.

Scaling does not have to be between -1.0 and 1.0 if the numbers used are not for sample values. The image information can also be scaled to MIDI numbers or to a frequency range and stored in an array to be used as a sequence or melodic fragment. This in turn creates a pseudo-algorithmic pitch generation process that reflects the changes in the pixel information. While the results of this process are different from those writing a buffer, the programming is very similar. In Figure 3.10, the first line of code loads the image into j while j.pixels gets all of the pixels and puts them into array o. The next line compiles all the image information and scales it to values between -1.0 and 1.0 and assigns it to p. Lastly, the scaled numbers are loaded into a buffer that can be played back. In the fourth line of the code, the same values are scaled to 0 – 127 (midi notes). Variable i's values will not be stored in a buffer but used as a string for control rate parameters.

```
j = Image.new("directory");
o = j.pixels

p = o.linlin(o.minItem, o.maxItem, -1.0, 1.0);

i = o.linlin(o.minItem, o.maxItem, 0, 127);

l = Buffer.sendCollection(s, p, 1, 0); \\loads
to buffer 0

{PlayBuf.ar(1, 0, rate: 1)}.play
```

*Figure 3.10 Supercollider Code for Scaling Pixel Information and Playing the Buffer*

These arrays vary in size and can be calculated by resolution X * resolution Y. This means that an image that is 3000x2525 pixels has an array of 7,575,000 entries, more than can be used in any reasonable fashion. 7,575,000 pixel values converted into samples would be a little less than 3 minutes (171 seconds) of sound if being used as a buffer. If the pixels were converted into frequencies and 1 frequency is played back every 1/1000 of a second it would take about 21 hours. This shows that there is too much information to use in an image without further filtering or restrictions. The most basic way to reduce the information is to change the dimensions of the image to X, Y <= 1000. If X is 500 and Y is 632 then there are 316, 000 pixels and would take 52 minutes to read as frequencies or have a buffer length of 7 seconds.

### 3.2.2 Using the Newly Interpolated Buffer

The buffer created from the pixel information is chaotic and changes over time. Sometimes the sound will be smooth and seem to evolve while at other times it is harsh and shrill. Occasionally, the sound takes more of a sawtooth wave shape and sounds as though it is producing a pitch with noise in the background. Regardless of the sound produced, it is stored in a wave table and traditional wavetable synthesis techniques can be used to manipulate the buffer.

The sounds produced by scaling the pixels between -1.0 and 1.0 are stored in lookup tables and have an index and sample value for the entire length of buffer. This allows for the table to be used just as a traditional lookup table is. The wave is bipolar and can be used as a modulation signal, or mapped to any other synthesis parameter. The sound itself can also be played at variable rates which will change the pitch and timbre of the sound or analyzed with FFT descriptors to obtain spectral information. There are many other ways of using the sound produced because it is no longer an image.

When the information is scaled to pitch or frequency it can serve as a melodic line generator. The pitch produced does not follow any recognizable pattern and images with high contrast (only black and white, no gray) will have a lot of very low frequencies, very high frequencies and not many mid-range ones. This creates a hole in the overall frequency range of the music and generally is uninteresting because the sound will only sustain a few very low and very high frequencies in a perceivably random rhythmical pattern. Frequencies that are below 20 Hz will also cause distortion to occur because when sustained, will modulate the frequencies being triggered. Pictures that have a wide variety of colors and shades will utilize the entire frequency range. More frequencies will also cause other effects to occur, such as phasing which in turn will cause the amplitude to vary in different frequencies. Repeated color in the image will cause repeated frequencies that can get triggered very quickly, producing a modulation effect until a new frequency is triggered. The frequencies being triggered can also be assigned to any type of generator, carrier frequency or synthesis technique. Figure 3.11 shows how a frequency argument is used to take an array and trigger a frequency back every 1/10 of a second.

```
j = Image.new("directory");
o = j.pixels
p = o. linlin(o.minItem, o.maxItem, 0, 127);

SynthDef ("image1", { arg freq;
var in;
        in = SinOsc.ar(freq, 0, 0.4);
        Out.ar(0, in);
})

r = Pseq(p, inf).asStream;
Task({
        inf.do({
                a = Synth.new(\image1, [\freq, r.next.midicps]);
0.1.wait;
};})
```

*Figure 3.11 Code for assigning generated frequencies to sound*

## Conclusion

Images can be turned into sound through a variety of processes depending on how the composer wants to use the information. Reading the image back as a sonogram can be used to create sound that can be analyzed for spectral properties and when plotted back still look like the image. The composer can then create a composition that reflects the changes in the image and properties of the picture. In contrast to this method, the image can be read and interpolated into a buffer to produce a waveform. This will create a sound that does not look like the picture when plotted through a sonogram, but will reflect the change in pixel information. This method reads from the top left to the bottom right and determines frequency by scaling the 128-bit pixel number to a range of samples or frequencies. In chapter 4, the image techniques discussed are applied to two compositions. The first is scored for an acoustic ensemble and the second for digital media.

# Chapter IV.

Spectral Compositions

## Introduction

In this final chapter, three compositions will be described and deconstructed detailing the processes involved. The first composition, titled *Establish, Corrupt, Broken* demonstrates, is a fixed media piece created using image information to generate waveforms and resultant spectra. It also shows the creative use of generating techniques. The second composition, *Laniakea Elegant Beauty,* shows how an image can be used to compose a composition for chamber ensemble. The last composition, *Invisible Signals*, illustrates how real-time spectral analysis can be used for score generation.

### 4.1 Fixed Media Composition

*Establish, Corrupt, Broken* is an 8-channel fixed-media composition that uses images of my life from the past six years. These images relate to important life events such as adopting a cat, getting a new car, studying with Anthony Braxton, working with Pauline Oliveros and more. Each image is shown in Appendix IV. Over the past 6 years, different cameras were used including iPhone 4s, iPhone 5s, Samsung Galaxy S3, Samsung Galaxy S6, a professional wedding photographer's camera, a Canon EOS Rebel T5, Canon 70-D and a MacBook Pro built-in camera. Each had different settings and maximum resolutions yielding pictures of different sizes, resolutions and qualities. This in turn changed the amount of data in the image and affected the resultant sound. Since the pixel information is not directly viewable, the various cameras create images with a certain level of uncertainty when transformed into sound. Not all of the image sounds were used in their entirety in the composition because of various timbral qualities.

### 4.1.1 Software Used for Image and Sound Processing

To turn the images into sound, each picture is first turned into a .png file and the pixel information extracted into a string of 32-bit numbers. The length of the string is dependent on the resolution of the image; i.e. higher resolutions create longer strings. Each number is then scaled to a value between -1 and 1 and stored in a buffer. A Supercollider SynthDef then reads the buffer as though it is a sound file, producing a sound that encompasses all pixel values read in order. For this composition, all of the parameters were kept the same between pictures in order to keep the picture information as intact as possible. The following Supercollider code in Figure 4.1 shows the translation from image to buffer to sound.

```
Server.default = s = Server.internal.boot

a = Image.new("/Users/maxwelltfirn/Desktop/ElectronicDissComp/car.png")

d = a.pixels
e = d.linlin(d.minItem, d.maxItem, -1.0, 1.0) //scale to sample number

b = Buffer.sendCollection(s, e, 1, 0) //send sample value to buffer index

b.plot //show buffer

//play sound
SynthDef("playback", { arg bufnum, rate = 1;
		var play;
		play = PlayBuf.ar(1, b.bufnum, rate, loop: 0);
		Out.ar(0, play);
}).add;

a = Synth("playback")

//record sound from server
s.record
s.stopRecording
```

*Figure 4.1 Image to Sound Translation*

This code also allows the synth to play the buffer slower or faster (denoted by the rate argument) which will affect the pitch. However, for the purpose of this composition, that argument is never changed from 1. All of the images are then recorded at a playback speed of 1. If a sound needed to be modulated, the modulation was done in a different program that will be described shortly. None of the sounds created in the above Supercollider code have an envelope, compression or limiting of the volume. All of the dynamic controls were done in Logic to avoid the constant recording of sounds with minute differences.

To modulate and temporally change the sound files, PVC written by Paul Koonce was used to transpose the sound, and then scaled by a certain factor. This gives more control over the transposition than does reading the buffer faster or slower. Apart from pitch shifting and time stretching, no other effects were used. Not only did this process create interesting sounds but also shows how diverse sounds can be without heavy processing. The composition, however, involves playing short samples back at a very fast pace, creating a pseudo amplitude modulation effect. This creates new sounds without actually processing the original sound. Short samples can also be interlaced to produce a variety of textures and timbres.

The goal of *Establish, Corrupt, Broken* was to create a glitch/noise composition rooted in spectral processing. While the sounds were created using synthesis, the rapid looping of short sounds is determined by various spectral properties. The spectrum was also used to further analyze the sounds for main frequency content in order to give them their own frequency space within 8 channels. This is very important because no equalization was used, and the goal was to keep the frequency relationships the same when transposing or stretching the sound. While this composition only spectrally shifts the sound, it also shows how spectral analysis can be used in mixing to balance the sounds within the overall composition.

Creating envelopes is very important because it gives each sound its own space in the mix and creates “pseudo-modulation” by rapidly changing the volume on short, fast sounds. All of the sounds of the second and third sections are derived from the first. The second section uses a hack-and-slash method, pasting sounds together and adding envelopes to create new timbres. Since all of the sounds are used in their raw form, envelopes are also used to control accents and intensities. Envelopes are normally used more for dynamic control rather than for timbre manipulation. Since I laid strict rules for what can be manipulated in the composition, creative uses of envelopes allowed for the rule to be bent while not actually altering the spectral content of the samples.

### 4.1.2 Form and Process

The form and process of the composition is directly related to those found in glitch visual arts. First an artist takes a picture/s (establishes information) and chooses which one to use. Then the picture is run through a process (data corruption) while still keeping remnants intact. Lastly, when viewing the new image, the original image is in it but takes on a new characteristic because of the glitch process. An example of this idea can be seen in Kate Wintjes's work titled *Contact Sheet.* In this work, a still video file of a woman is fed through a system to produce glitches in the image's information. However, within each glitch the original image of the woman is still recognizable.

The form of the composition is ABC in terms of material, but AB in relation to the techniques employed. The first section, *Establish*, uses long sounds that overlap without clashing in frequency space. *Establish* is also a metaphor for the creation of information. The envelopes are long, and are trapezoidal or triangular, with long fade-ins and fade-outs. *Establish* does not have much silence but uses texture to create shifting densities and volumes. The maximum density uses all eight channels while the minimum uses only one. The density also shifts between high frequency sounds that sound less dense to low sounds that feel denser. By incorporating the perception of density based on frequency content as well as actually density, there is a lot of room to manipulate the sonic atmosphere.

The second section, *Corrupt,* uses a different process to create a consistently thick and violent texture. All 8 channels are used, with each channel containing very short samples derived from the sounds of section 1. These samples range in length from 3 milliseconds to 1 second. Various samples from different sounds are used in this process to generate new sounds that change very quickly and sometimes create rhythms. Each sample also has a separate envelope, resulting in various amplitude modulation effects within each sound. Each second of sound for one channel in this section has between 1 and 70 sound samples. That means the maximum amount of sound from all of the channels will have 560 sound samples per second. This not only creates a very dense texture but also allows any sound with a different spectral range to naturally stand out from the other sounds. This section also rapidly changes sounds between speakers so the texture travels in 3 dimensions while not changing the overall density of the atmosphere. The spatialization allows for sounds to be separated that would normally be in the same area of the spectrum and sound muddy. The separation also enables more sounds to be used without overlapping frequencies, thus creating thicker textures. The overall mixing technique of this section can be described as a "wall of sound"[iii].

*Corrupt* is a musical metaphor suggesting change of information in computer systems and directly contrasts with the idea of *Establish.* Here, the process takes the "pure" sounds from section 1 and "corrupts" them to make them unrecognizable. In computer glitches, there are still remnants of the original data in the corrupted file. In this section the corruption makes the original audio files unrecognizable while keeping all of the spectral information of the original sounds.

The final section, *Broken*, suggests a musical struggle between the original "pure" sounds of the beginning and the "corrupted" sounds. This idea is taken from a destructive glitch finding in which the original source is destroyed in the process of finding the glitch. This also happens in destructive sound editing when the original sound is altered in the software and rewritten upon saving. This last section uses a rebuilding process and attempts to reconstruct similar characteristics to those in the first section using remnants of the second. The sounds themselves could have been copied and pasted from the beginning to the end or the names of the files could have been viewed and samples could have been pasted back to together. However, since destructive editing does not normally allow for copy and paste, I looked at the waveforms and tried to piece the sounds back together. Some sounds have a similar quality but on a whole I

[iii] Wall of Sound refers to a technique for mixing audio created by Phil Spector. For further reading about Phil Spector's "Wall of Sound" technique refer to Brown 2007

failed to reconstruct a sound from the beginning by pasting small clips together using visual information. This failure creates a quiter section with spectral subtleties standing out more than they do in other sections. This type of failure also allowed for the last section to relate to the first section while still having its own properties.

*Broken* involves silence, used much more than in the earlier sections. Silence provides the listener with possibility of moment-to-moment contemplation and the time to recognize familiar material. The piece ends very quietly with sounds heard from the beginning as a way of uniting the sections of the composition and showing that the sounds never actually changed.

*Establish, Corrupt, Broken* fuses timbre and spectral considerations with image-to-sound processing and with noise/glitch music. While this composition does not use spectral manipulation, I used spectral analysis to view detailed parts of spectra for mixing and balance purposes. While spectral music is usually regarded as a process for creating music, in this composition I use it as a balance guide for studio mixing.

### 4.2 Chamber Ensemble

*Laniakea Elegant Beauty* is scored for flute, clarinet, 2 violins, viola, cello and double bass and uses an image of the Laniakea Supercluster as its spectral source. The image of the supercluster is reminiscent of a cobweb or jelly fish. There is a bright central band that consists of more densely packed galaxies while thinner filaments extend outwards into a web like structure. The image has a natural visual hierarchy with the bright band standing out more than the filaments and the filaments are in front of background stars. This hierarchy translates into a musical spectrum after the image is converted to sound.

### 4.2.1 Compositional Process

To obtain information from an image of the Laniakea Supercluster I loaded it into my image-to-sonogram program described in Chapter 3. The brightness of the color directly relates to the volume of the individual frequencies. This in turn allowed for the filtering of soft sounds or noise from the image and maintenance of the most prominent features. Once the sound was filtered I broke it into 10 equal parts and converted each into an sdif file. These files contain all of the onset, frequency and amplitude information of the sound and can be parsed to obtain a variety of types of musical information. In Figure 4.2 the image and the resultant spectral image is shown.

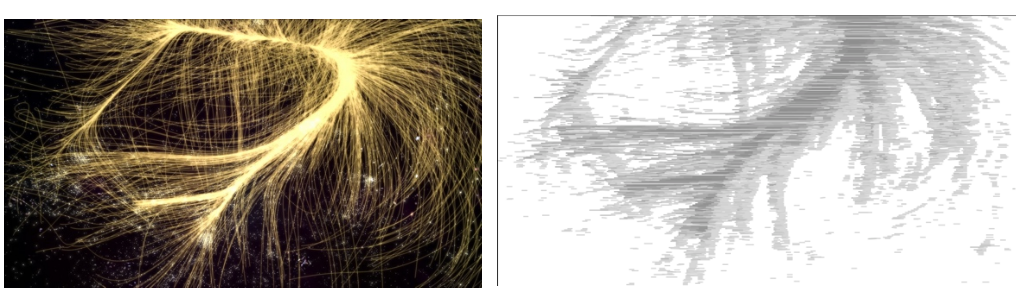


*Figure 4.2 A) Visualization of the Laniakea Supercluster, B) Filtered Spectrum of Sound*

Each section is then run through my Image Score software that parses the onset and frequency information of the sdif file and notates it on a music staff. This gives a notational view of how the sound changes over the length of the section. Amplitude is not used for this because the ultimate goal of this process is to obtain harmonic information over time that will anchor the compositional structure. The composition then takes its form from the harmonic progression while the rhythm, dynamics, transitions and timbre are freely composed.

The spectrum resulting from this process is aperiodic and changes very quickly, unlike that found in naturally occurring sounds. Because of this quality, not all information can be taken from the spectrum in a way similar to that used by Grisey or Murail. Instead, the composition uses the spectrum as a guideline similar to its usage by the Romanian spectral composers. This creates enough freedom to transition between sections of the image smoothly and to compose using the harmonic progression. The composition was originally sketched to be spatial in time and uses quantization to fit into standard score notation. This allows spatial properties to be represented while keeping consistency between performances.

### 4.2.2 Form

The image is denser towards the middle with the outlying filaments being sparser. To reflect this in the music, the composition begins with long tones in the instruments with periodic disruptions of short quick flurries. This creates a thin atmosphere with slight variance just like the filaments in the image. Dynamics in this early section are also on the soft side because the filaments in the image are not the direct focal point. Short, fast flurries are interjected into the long tones to both represent the stars in the background of the image and cause movement in the overall sound.

As the composition progresses, it approaches the denser center band of the image. To represent the image musically, long tones get shorter and shorter until they become fast poly-rhythmical passages. The curve of the central band makes a shape similar to a backwards 'C'. The curve of the 'C' has filaments coming out of the band and spreading out. The instruments at this point periodically rest or move to long tones for variable lengths of time. The curve also is used for the music to transition between a densely packed section (the main curve is dense and bright) back to a long, drone section over time as shown in Figure 4.2.

The last section of the composition transitions from the middle, dense, rhythmical part to one that is similar to the first section. This represents the image as well as allowing for the atmosphere to change. In the image, the right side has a lot of thin filaments that are not in the foreground but still show varying degrees of density. Musically, this is achieved by using sustained sounds but also the fast repetition of notes. Sustained sounds create a long, light atmosphere while the repeated sounds add static density to the overall sound. This creates music that feels light but at the same time has more density to it than the first section.

### 4.2.3 Techniques

The composition is mostly in 4/4 meter at 60 beats per minute in order for the spatial quantization to work. This enables rhythm and gesture to be notated with consistency. The instruments use various techniques to represent the image and certain aspects of the spectrum. The flute uses flutter-tonguing to create a sense of motion in static notes while the bass clarinet uses multiphonics to add density to the ensemble's sound and to represent noise and instability. In the strings, *pizzicato* is used to create a granular sound, representing the stars that comprise the supercluster as well as harmonics to achieve higher-than -normal frequencies in the spectrum. Further, *sul ponticello* is used to bring out the higher harmonics in the strings.

Using these techniques allows variations in texture as well as a more accurate representation of the spectrum. While some characteristics of the image do not have a direct relationship to music, certain musical ideas can suggest elements of the image, such as stars represented by *pizzicato* notes. Dimensions in the image are used to create texture, with small lighter filaments captured by soft long tones while density and brighter colors layer sound at louder volumes.

*Laniakea Elegant Beauty* is meant to explore compositional possibilities when interpreting and using an image as a sound spectrum. This exploration is different from that of composers such as Grisey or Murail because a new layer of information is added to the compositional process. How to represent the spectrum and what elements are important is constantly considered, rather than basing the entire piece on one spectral element. Within this framework, I used obvious color differences and observable densities to make decisions, however there is no established way of doing this. Another composer could take this framework and decide to use the harmonic content, the image's characteristics, or both. Each way of thinking can create a variety of musical possibilities.

### 4.3 Real-Time Spectral Composition

*Invisible Signals* is a composition for open instrumentation, electromagnetic microphone and electronic toys or devices. The score is generated by a performer using an electromagnetic microphone to pick up electromagnetism from the electronics. The microphone then turns the magnetism into a line level electrical current i.e. sound. The sound is then analyzed in Max/MSP for the 16 loudest frequencies and notated in order from lowest to highest on a music staff. This "scale" gets refreshed every 100 milliseconds in order to pick up all of the subtle changes in the incoming sound.

During the composition, the performers read the notes produced and improvise with them while trying to match the quality of the electronic sounds. It is performers job to make decisions about if their playing does not fit the sound or ensemble and change accordingly. Performers also have the option to play in contrast with electronic sound. However, all performers are not allowed to play in contrast and the contrasting part must be in the background. By adding this rule at least one performer will always match the electronic sound and the contrasting parts will add texture and timbral differences.

### 4.3.1 Software

The computer part is created using Max/MSP, the IRCAM FTM library and the Bach Automated Composer Helper library. The software is divided into 2 parts, sound analysis and the notation. By dividing the software into parts, changes to sound analysis and notation can be separated and allow for different resolution changes.

The sound analysis part of the program uses the FTM library to break apart the sound and obtain the FFT information. The incoming sound is broken down into frames of 1024 samples and enveloped using a Hanning function. The enveloped window is then sent through an FFT analysis with a frame size of 8192 to optimize frequency resolution. The FFT information is then converted into a logarithmic, absolute value number and sent to gbr.peaks. gbr.peaks analyzes the incoming information for the peak values based on user controlled parameters and outputs the frequency and amplitude. The internal parts of the program can be viewed in Appendix II. As part of the analysis, spectragraphic information is shown so the performer can see the sound and the other frequencies that are not peaks.

Using the Bach library, the information from the gbr.peaks analysis can be converted into cents and notated in time. The list of information for gbr.peaks is a 2 x N matrix with frequency and amplitude represented. The list is divided so only the frequency in hertz information is used and every 100 milliseconds an FFT window is converted into usable information. After the conversion is done the information is transposed down by one octave to allow more notes to be seen by the performers without having to count ledger lines.  The transposition allows the performers to efficiently read the notes.  Without this many notes have too many ledger lines to read quickly in real-time.  Normally the frequencies of the sounds will have an onset.  However, because the sample rate for notation is being controlled by the computer player, the pitch onset is not used and any sustained peak frequency will show up in each refresh of the notation.

### 4.3.2 Form

The form of *Invisible Signals* will change in every performance because there are no instructions for moving the microphone between toys except to transition slowly.  The form is produced in real-time by the performer controlling the electronics.  This allows the ensemble playing the composition to figure out a form and pace to pre-compose the microphone transitions.  The ensemble can then choose to keep that form and pace between performances or change it every time.  However, the performer can also choose to improvise everything about the composition in real-time.

The purpose of leaving the composition open is to highlight the unstable nature of electromagnetic sounds.  Some sounds are subtle with a lot of detail that the performers may want to expand upon while others are more aggressive or harsh and a shorter time for them may be determined.  While the composition could be scored or use prerecorded sounds for consistency, having the sounds formed in real-time creates unpredictability that can be reflected in the music.  The sound analysis also takes into account any other electromagnetic waves in the air giving the composition another level of uncertainty.  The refresh rate of the notation is too fast for any performer to keep up with so a constant shift of pitch, timbre, dynamics and rhythm will occur throughout the composition. The performers act as an information filter as each performer is filtering the information at different rates. This creates natural transitions between instruments, which occur at a pseudo-random pace, and shifting textures.

*Invisible Signals* takes traditional French spectral ideas and fuses them with the experimental qualities of the Romanian composers and the techniques of computer aided composition. The French spectral school is represented by the musical information being taken from only the spectrum and used in a "set" like fashion harmonically. The Romanian school's experimental nature is seen through the improvisatory nature of rhythm, dynamics and techniques. The second generation and newer spectral thought is found in the real-time score generation, and computer aided composition. While this composition takes small ideas from the different spectral schools, it shows how a new approach can be taken from already established ideas.

**Conclusion**

The three compositions described above are intended to show how spectral approaches can be used in different styles of composition. The first uses the image-to-sound processing to generate timbre while using spectral information to guide mixing, while the second composition uses an image to generate frequency content and form. The last composition illustrates a real-time spectral composing possibility. These compositions are meant to fuse the foundational work of Murail and Grisey with the experimentation of Dumitrescu, second generation spectral composers and glitch art.

## Epilogue

Spectral music and sound manipulation have a close relationship built on the analysis and realization of a sound. The fundamental spectral approach created by Grisey and Murail and the scientific study of timbre laid the foundation for a new compositional practice that is still used in many genres of music. Using the established foundation to develop my work, I have examined how spectral music has evolved and diverged in a variety of different musical forms and techniques.

At the start of this research, I believed that spectral music had a particular sound and aesthetic that needed to be followed to be accepted as such. However, I discovered that it is a broad set of compositional practices and interests. By studying the compositional processes of composers such as Grisey, Murail, Dumitrescu, Saariaho and others, I learned various techniques used for timbre-oriented composition. By illustrating these composers' ideas and creating software that crosses disciplines, I have shown how to incorporate different spectral practices using a variety of new techniques.

As my work continued, I explored pop culture music such as Aphex Twin, Venetian Snares, Merzbow and Ryoji Ikeda. These artists show an extreme interest in timbral manipulation. However, many of their synthesis techniques and musical practices have parallels to those of spectral composers. This idea is contrary to Hugues Dufort's definition of spectral music but is connected to Grisy's "liminal" music. The original spectral manipulation techniques and later spectral practices served as the inspiration for the new techniques and compositional processes in this study.

The software and three compositions created for this dissertation fuse new spectral ideas that are focused on the analysis of timbre and interdisciplinary practices as well as electronic, acoustic and improvised music. This research has shown that spectral music is not a sound or process but a compositional interest in timbre. While historically, analysis and manipulation were extensively used in spectral compositional practices, I believe it is really the interest in the nature of sound that pushed spectral music to where it is now.

My three compositions illustrate different approaches to spectral music. *Establish, Corrupt, Broken* was inspired by noise and glitch art but focuses on the source of the timbre (images) and how the timbres change over time. *Laniakea Elegant Beauty* and *Invisible Signals* were both inspired by analysis based spectral techniques but take a modern approach for the sound source. *Laniakea Elegant Beauty* uses an image as a sonogram and analyzes the harmonic progression while *Indivisible Signals* uses analysis in real-time to create an improvisatory score. These compositions enabled me to incorporate my love for noise music and experiment with the interpretation of images into a new spectral approach. While two of the three compositions use images as initial sources, they utilize different approaches that can also be incorporated outside of image to sound processing. The image techniques created have a foundation in spectral music and can be used for sonification practices as well as process based music composition.

**Future Study**

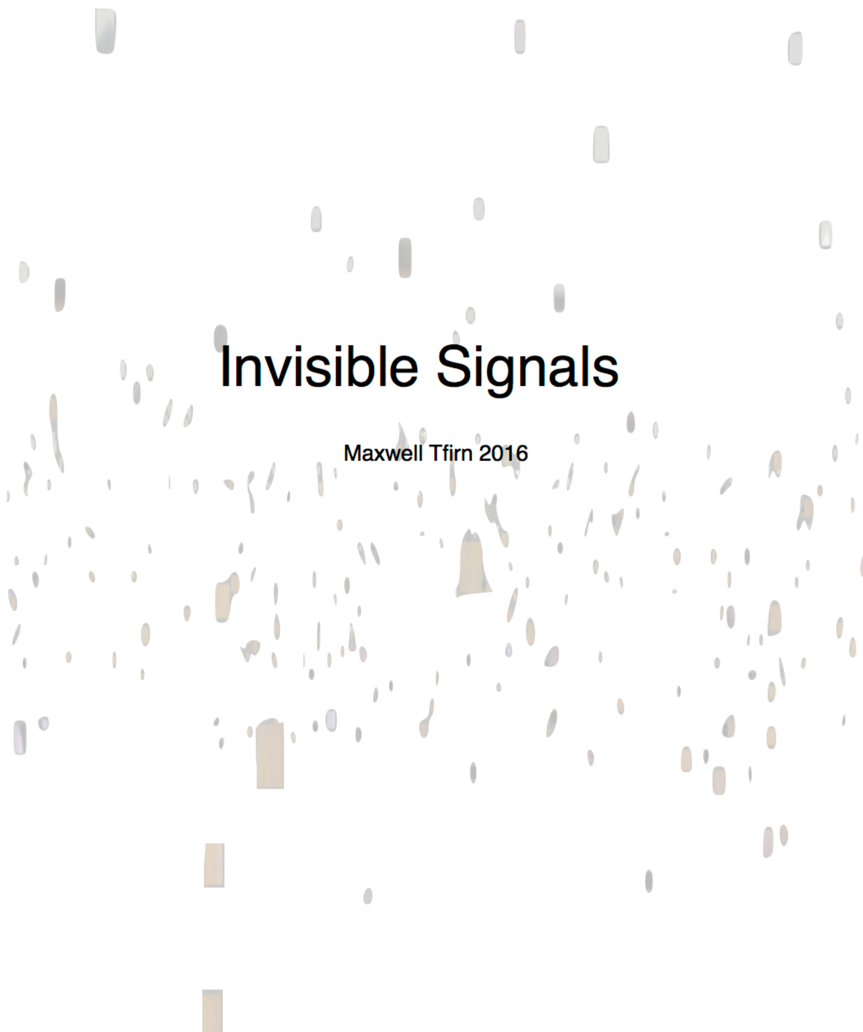
Future directions of research include the use of additional spectral analysis types, including mel bands, bark scales and attack estimation. While the descriptors in this dissertation are used for music composition, others could also help explain how the spectrum is perceived or deal with asymmetry of the spectrum (Peeters, 2004, p. 13). Other descriptors can also be used to create MFCC (Mel frequency cepstral coefficients) calculations for linear predictive coding or speech recognition techniques (Peeters, 2004, p. 5).

In the future, the reverse processes of taking a sound and turning it into an image can yield another technique for data manipulation. I would like to develop software that interpolates a sound buffer into an image as a way to create new abstract images for analysis. The image produced from this process will not describe frequency and time as in a sonogram, but instead use the raw data to make abstract images out of the sound. Compositionally, the image could be used to create a new timbre if run through the software made for this dissertation, so additional levels of abstraction can be added to the sound and data generation process.

In addition to using or creating images, I would like to develop a more detailed approach for the generation of real-time spectral scores. Real-time notation can also generate scores from complex improvised performances. By using spectral analysis in real time with detailed score generation, a composer or performer can improvise music and have a score generated that they can then edit into a permanent composition or analyze or to learn about their improvisational trends.

Lastly, I would like to explore neural networks and use them to learn spectra and create new ones. By training a neural network using complex spectra, the neural net will identify patterns that are not discernable to humans. Then the neural net will create a new spectrum that is completely artificial that can then be used for music composition and sound manipulation. While the product of the network cannot be unknown in advance, it offers endless timbral possibilities by simply retraining the network with a variety of sounds. The neural network can also be trained with similar sounds such as various samples of snare drums, as I have already done. The network in turn can create a sound similar to that of a snare drum, but with different "unnatural" characteristics. Since the acoustic sound is computer generated, the neural net can also be used to manipulate it and create an "acoustic" sound that is not possible with the given instrument. This process can also be used to generate unique electronic sounds by retraining the network with the appropriate data. All of the processes using neural networks will generate new sounds to be used in electronic compositions or spectra that can be analyzed and used for creating acoustic spectral music. This technique as well as the others described above offer a path forward in the rich domain of spectral music.

**Appendix I: Invisible Signals**



**Program**
Invisible Signals is a composition that explores the unheard and unseen electromagnetic waves that surround us. During the composition these waves are captured by an electromagnetic microphone and sent as a line level signal into a computer. The computer then analyzes the signal for the 16 loudest frequencies and notates them as a scale in a staff. The performers then use this scale as well as a sonogram to improvise and match the electronic timbre.

**Ensemble**
The ensemble can be any instrumentation with at least five members. Six members if the performance is interactive. Any instrumental technique can be used as long as it reflects the electronic sounds and blends with the ensemble. The duration of the performance should be no less than 10 minutes.

**Equipment**
Computer, Stereo Sound System, Max/MSP, Projector, Screen, Either an Electromagnetic Microphone or Supplied Electronic Part. Setup instructions are below.

**Structure**

| | | |
|---|---|---|
| Soft<br>Sporadic<br>Blending | Loud<br>Build To Longer Tones<br>Mixed Short Attacks | Soft<br>Deconstruct |

The Above Structure is a suggestion. The ensemble can choose a new structure or if the performance is using a mic, match the electronic sound that is being produced in terms of atmosphere, density, texture and dynamic.

**Software**
If using the provided sound file:
- Load the sound file into the software and push play to start.

If using a Microphone
- Connect mic to computer and in the audio settings change the input source to the interface and built-in input.


Click the Click First Button then click the Click Second Button, both outlined in **RED**

If using a sound file click the Load Sound button. The play button will start the file. Outlined in **BLUE**

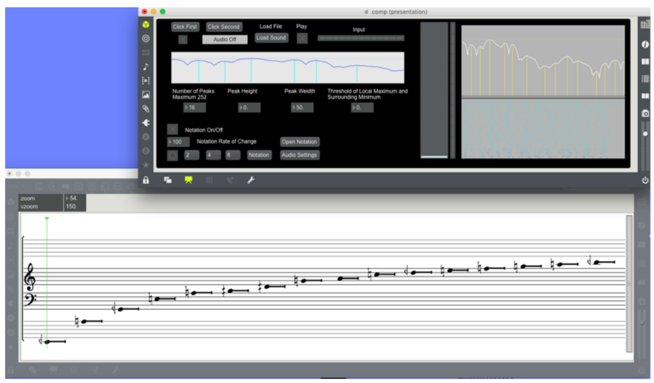
Slider Outlined in **Purple** controls master volume

Notational Rate Change outlined in **Yellow** controls the speed of scale refresh. For the performance it should be 100. For rehearsal it can be slower to allow performers to adjust.

Click the Open Notation Button before starting the composition.

If a real-time performance, click the Click First and Click Second buttons. Raise the volume slider, click the open notation button and click the notation on/

The below images shows how the projection should be set up for the performers. Apart from the notation on the bottom, the spectrogram and sonogram on the upper right side should be viewable. These are a guide for the performers and give insight to what the spectrum looks like outside of the analysis.

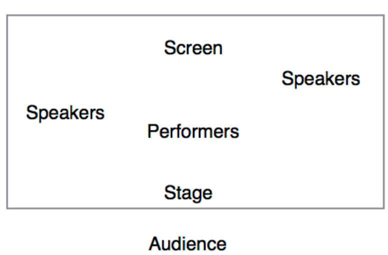


**Note**

For interactive performances one performer should have a table of electronics and the microphone. The performer then interacts with the electronics at their own pace. This performers sets the form of the composition by how they interact and transition between sounds and electronics.

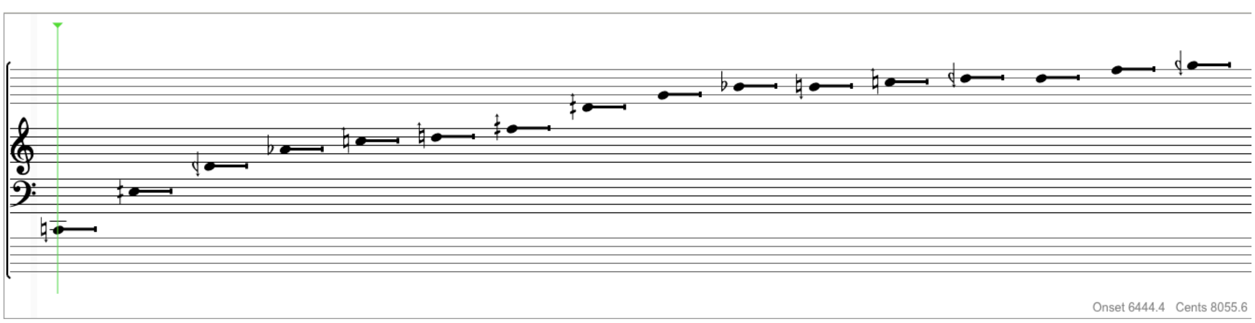
**Setup**

All performers read from the projected screen with their backs facing the audience

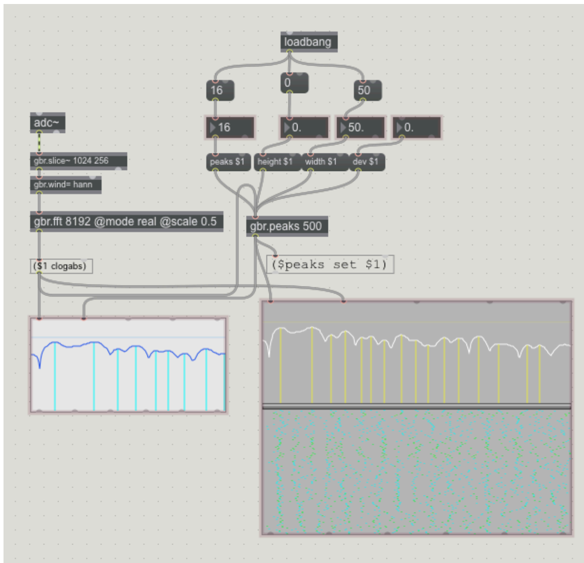


## Appendix II. Invisible Signals Software

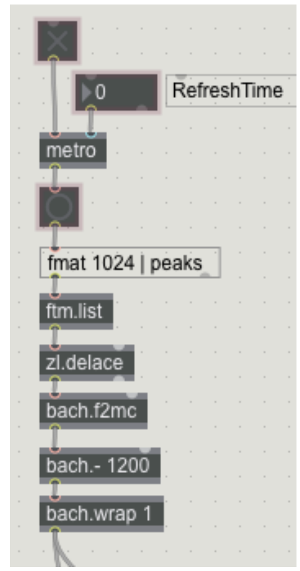

Main Screen




Score Screen




Analysis




Score Generation

**Appendix III: Laniakea Elegant beauty**

# Laniakea Elegant Cluster

Maxwell Tfirn


8
Fl.
B. Cl.
Vln. I
Vln. II
Vla.
Vc.
D.B.
mf
fp
mf
n
mf
p
mf
ppp
f
p
sul pont.
mf
p
11
Fl.
B. Cl.
Vln. I
Vln. II
Vla.
ord.
Vc.
D.B.
mf


Fl.
B. Cl.
Vln. I
Vln. II
Vla.
Vc.
D.B.

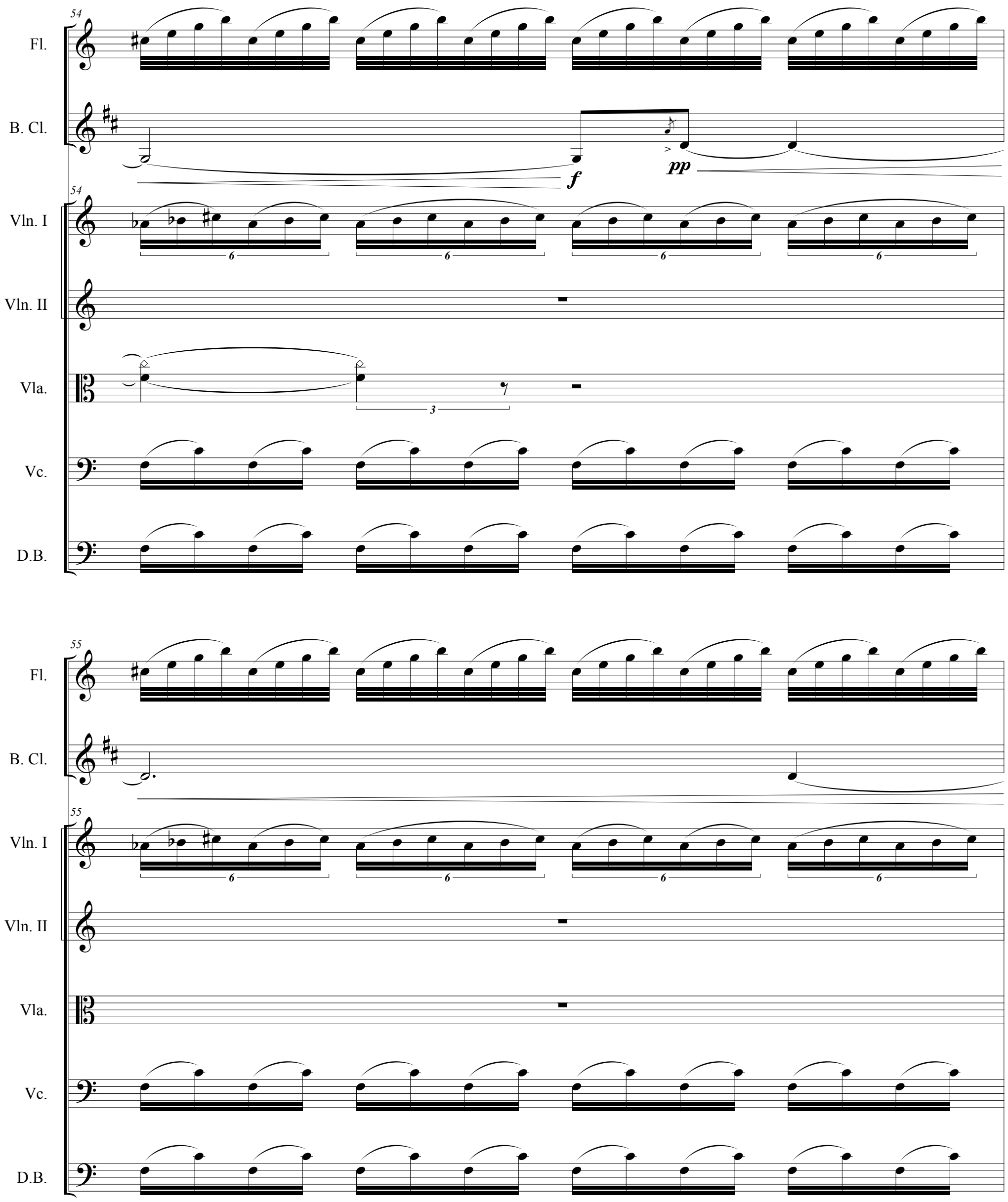

19
Fl.
B. Cl.
19
Vln. I
Vln. II
Vla.
Vc.
D.B.
25
Fl.
B. Cl.
25
Vln. I
Vln. II
Vla.
Vc.
D.B.
fp
mf
p
mp
mf
mf
Play Behind Bridge
mf
mf
n
n
mf


Fl.
B. Cl.
Vln. I
Vln. II
Vla.
Vc.
D.B.


39
Fl.
B. Cl.
Vln. I
Vln. II
Vla.
Vc.
D.B.
42
Fl.
B. Cl.
Vln. I
Vln. II
Vla.
Vc.
D.B.

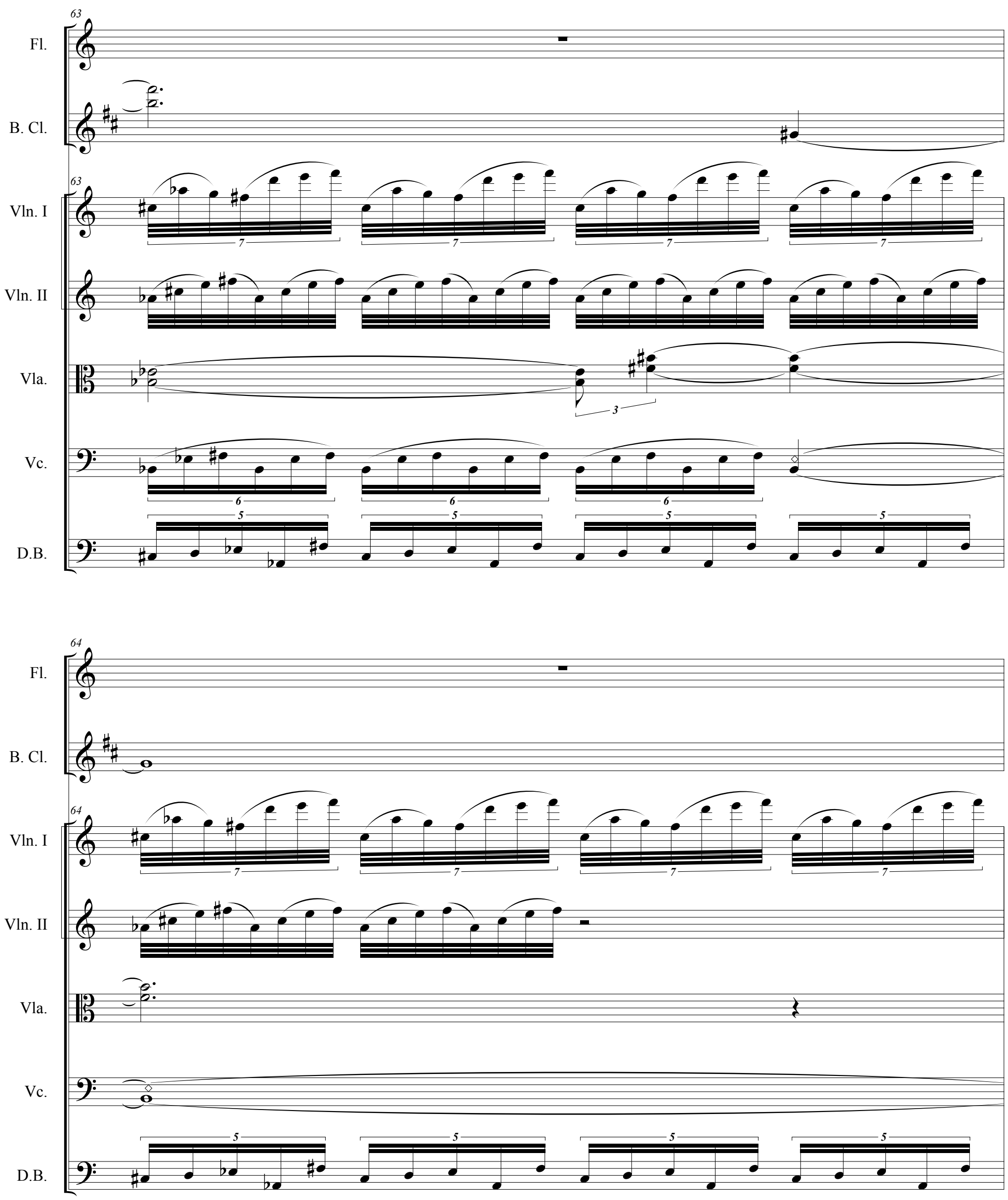

45
Fl.
B. Cl.
Vln. I
Vln. II
Vla.
Vc.
D.B.
Play Behind Bridge
sul pont.
ord.
pp
48
f mf
pp
f


50
Fl.
B. Cl.
Vln. I
Vln. II
Vla.
Vc.
D.B.
f mf
sul pont.
mf
52
sul pont.
pp
sul pont.
sul pont.
pp

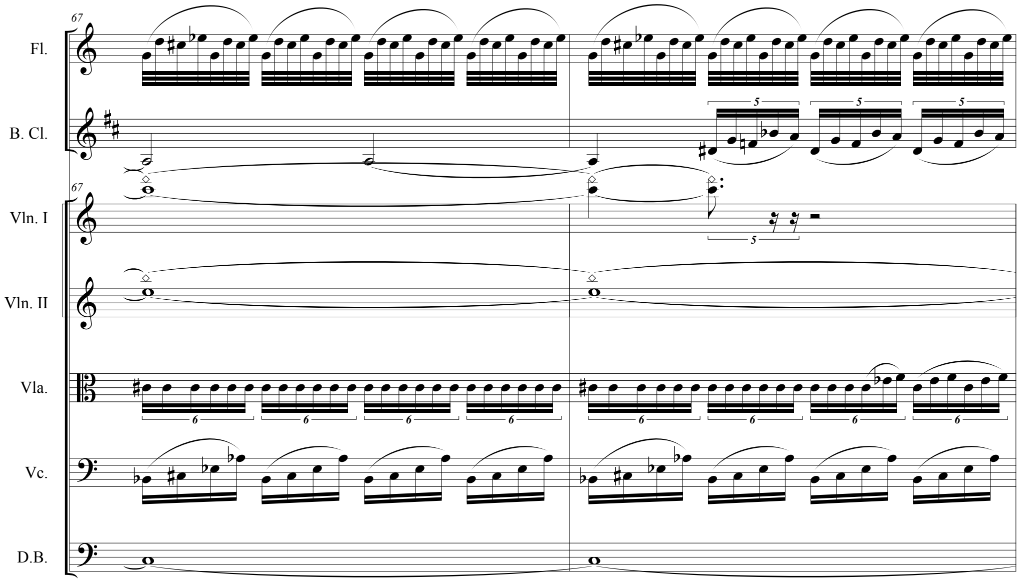

54
Fl.
B. Cl.
f
pp
54
Vln. I
Vln. II
Vla.
Vc.
D.B.
55
Fl.
B. Cl.
55
Vln. I
Vln. II
Vla.
Vc.
D.B.

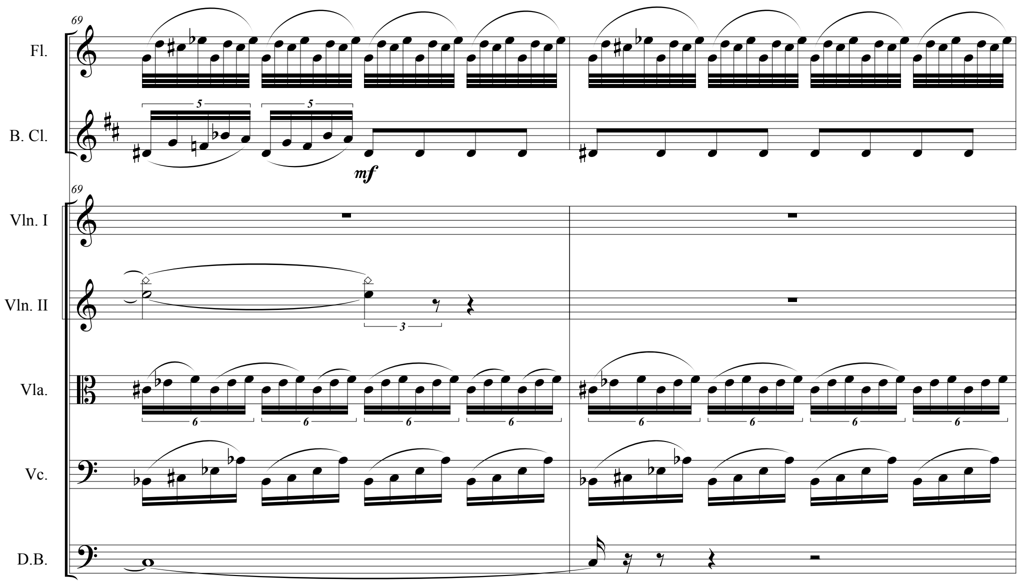

56
Fl.
B. Cl.
Vln. I
Vln. II
Vla.
Vc.
D.B.
57
sul pont.
mf
pizz.


59
Fl.
B. Cl.
Vln. I
Play Behind Bridge
Vln. II
Vla.
Vc.
D.B.
61
Fl.
B. Cl.
Vln. I
Vln. II
Vla.
Vc.
D.B.


63
Fl.
B. Cl.
Vln. I
Vln. II
Vla.
Vc.
D.B.
64
Fl.
B. Cl.
Vln. I
Vln. II
Vla.
Vc.
D.B.

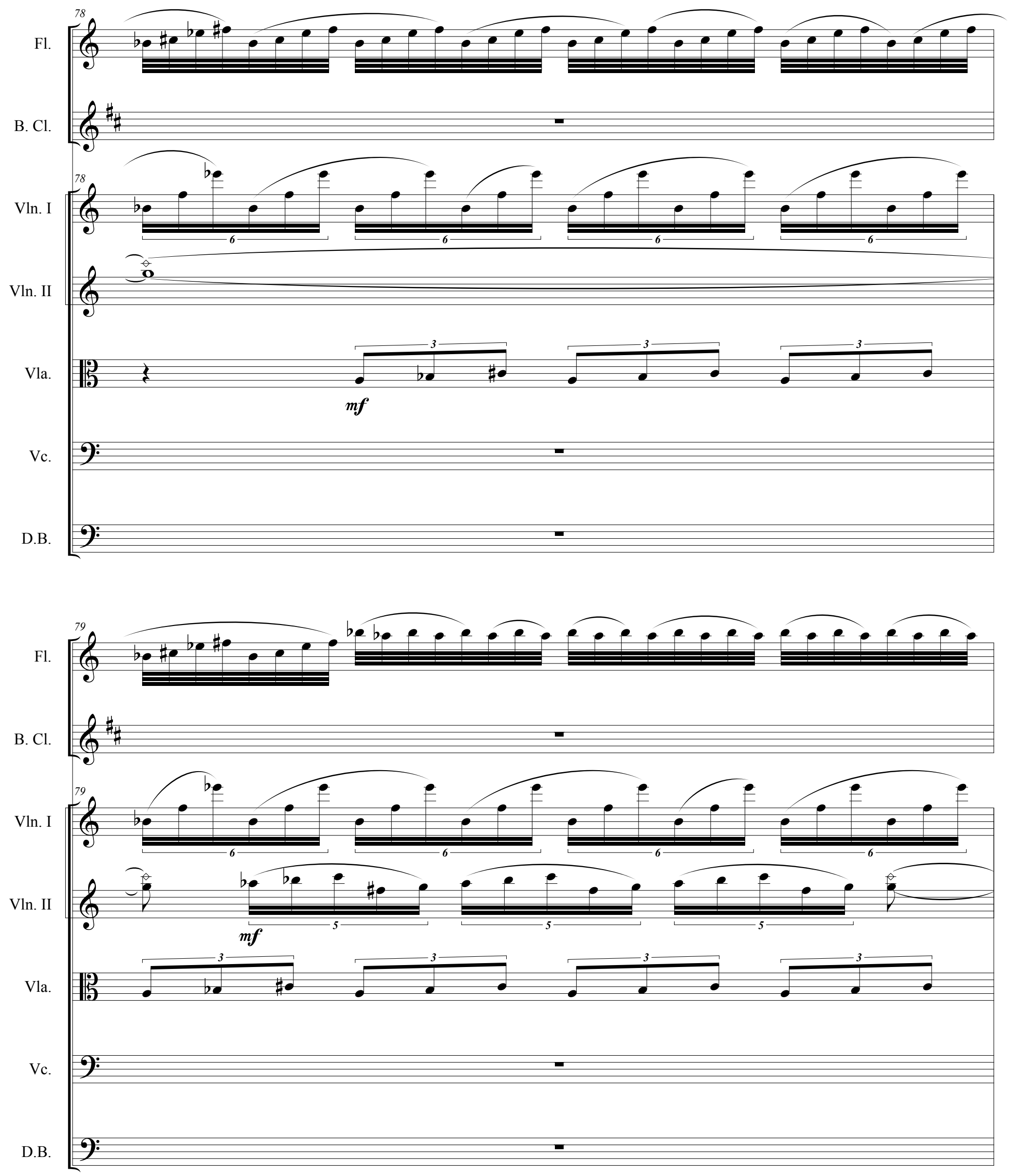

65
Fl.
B. Cl.
Vln. I
Vln. II
Vla.
Vc.
D.B.
66
arco


67
Fl.
B. Cl.
Vln. I
Vln. II
Vla.
Vc.
D.B.


69
Fl.
B. Cl.
mf
Vln. I
Vln. II
Vla.
Vc.
D.B.


71
Fl.
B. Cl.
71
Vln. I
Vln. II
Vla.
Vc.
D.B.
73
Fl.
B. Cl.
p
73
Vln. I
Vln. II
Vla.
p
Vc.
p
D.B.
p


Fl.
B. Cl.
Vln. I
Vln. II
Vla.
Vc.
D.B.
f
Fl.
B. Cl.
Vln. I
Vln. II
Vla.
Vc.
D.B.
mf
mf
p


78
Fl.
B. Cl.
Vln. I
Vln. II
Vla.
mf
Vc.
D.B.
79
Fl.
B. Cl.
Vln. I
Vln. II
mf
Vla.
Vc.
D.B.


80
Fl.
B. Cl.
Vln. I
Vln. II
Vla.
Vc.
D.B.
mf
82
Fl.
B. Cl.
Vln. I
Vln. II
Vla.
Vc.
D.B.
mf


84
Fl.
B. Cl.
Vln. I
Vln. II
Vla.
Vc.
D.B.
p
f
p
p
p
p
p
86
Fl.
B. Cl.
Vln. I
Vln. II
Vla.
Vc.
D.B.
mf
mf


89
Fl.
B. Cl.
89
Vln. I
sul pont.
mf
Vln. II
sul pont.
mp
Vla.
sul pont.
mp
Vc.
mf
pizz.
D.B.
mf
91
Fl.
B. Cl.
91
Vln. I
Vln. II
Vla.
Vc.
D.B.


93
Fl.
B. Cl.
Vln. I
Vln. II
Vla.
Vc.
D.B.
mp
p
arco
95
Air Only
ppp
f
pizz.

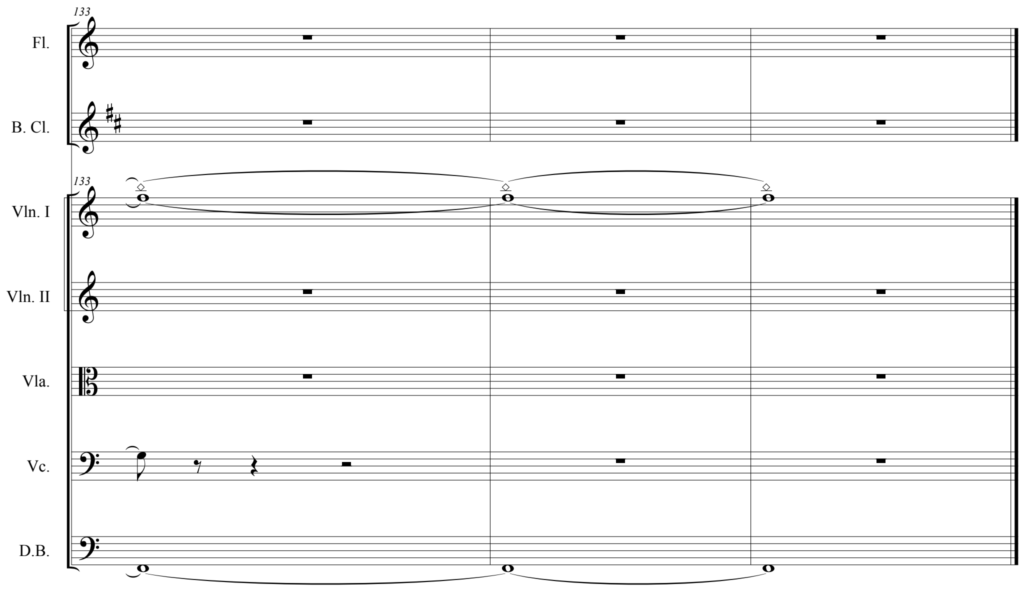

97
Fl.
B. Cl.
Vln. I
Vln. II
Vla.
Vc.
D.B.
p
mf
arco
99
ord.
pizz.


101
Fl.
B. Cl.
Vln. I
Vln. II
Vla.
Vc.
D.B.
3
p
mf
6
103
arco


105
Fl.
B. Cl.
Vln. I
Vln. II
Vla.
Vc.
D.B.
mp
109
p
n
sul pont.

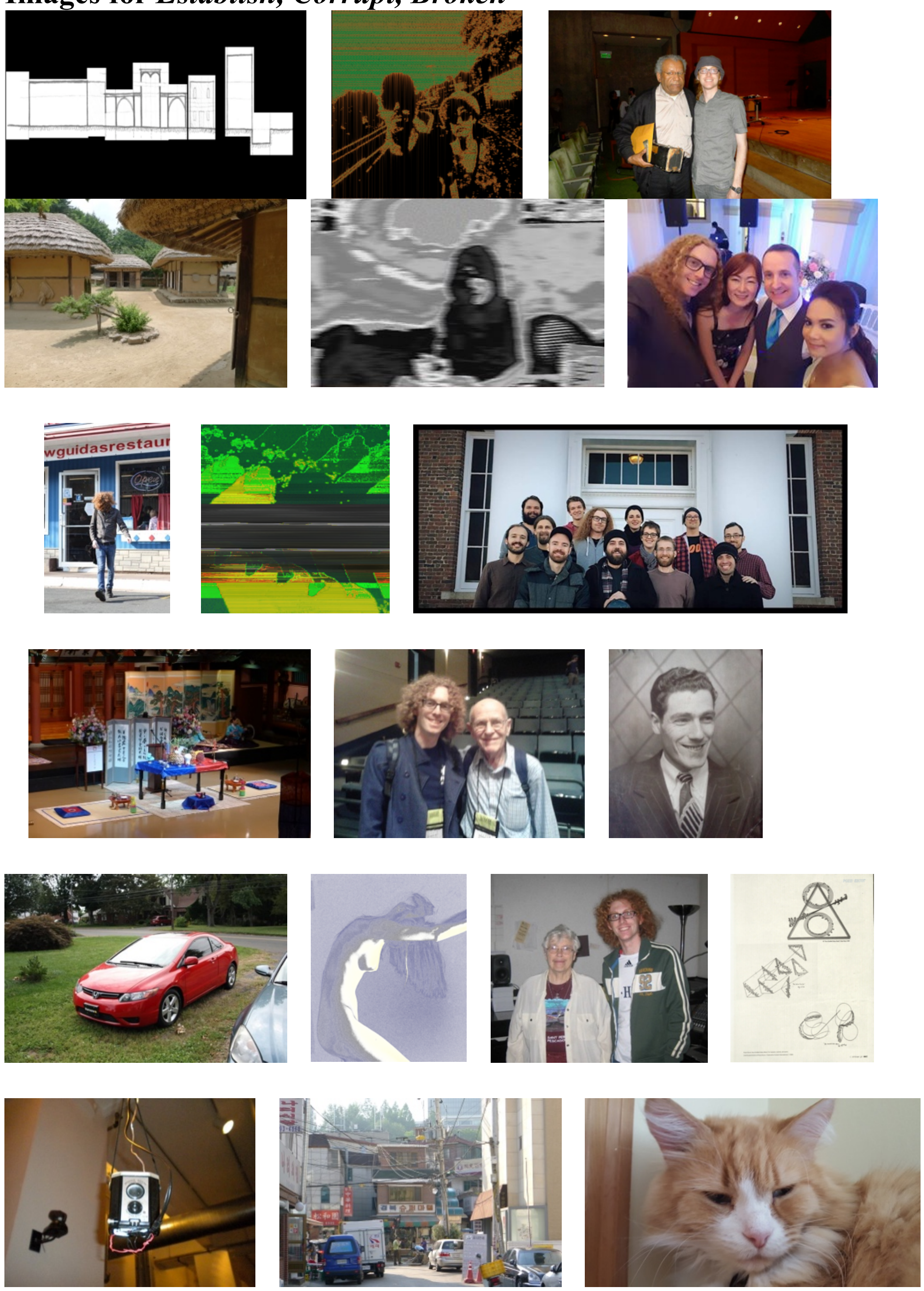

116
Fl.
B. Cl.
Vln. I
Vln. II
Vla.
Vc.
D.B.
sul pont.
ord.
pp
124

Laniakea Elegant Cluster



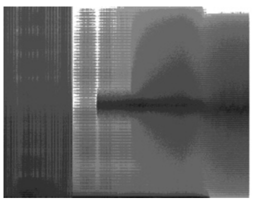
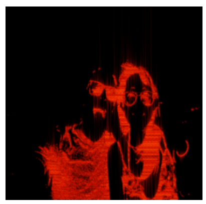
## Appendix IV. Establish, Corrupt Broken Images

**Images for *Establish, Corrupt, Broken***







# Appendix V. Additional Software

## Image to Sonogram

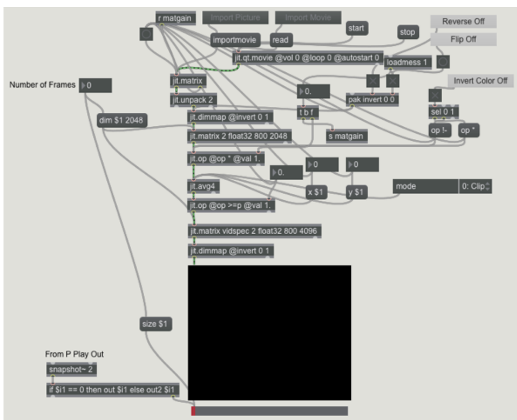
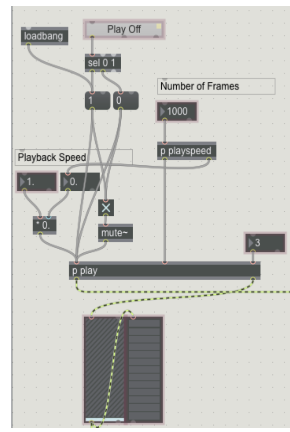
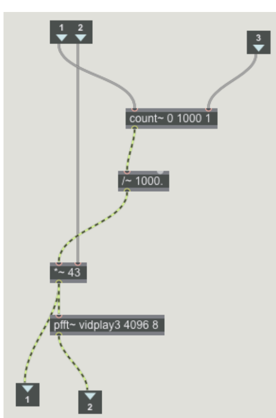


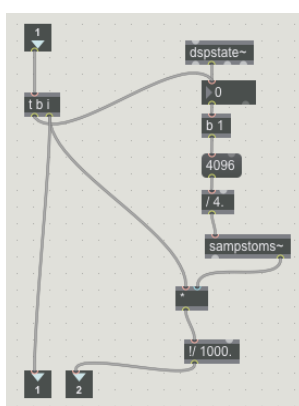

Image Data Rendering



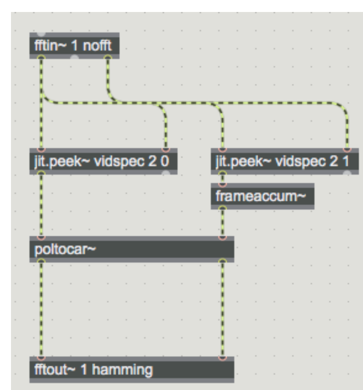

Audio Processing




P Play Subpatch

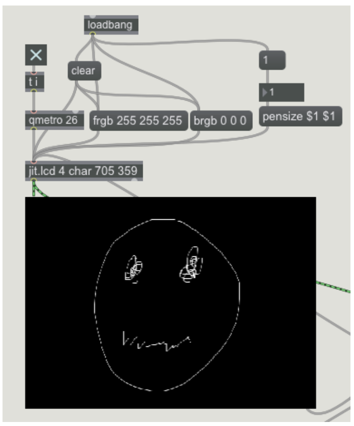



P Playback Subpatch

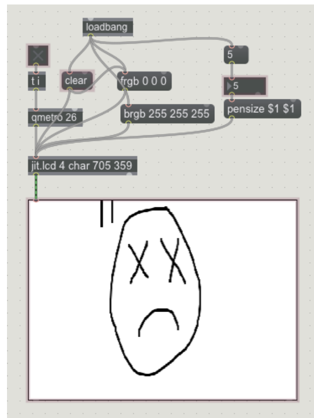



Vidplay Subpatch

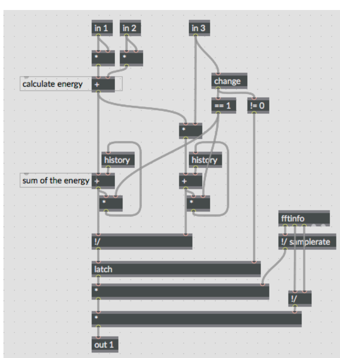
**Image Draw**



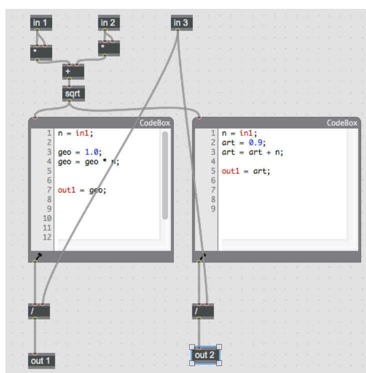

**Erase Spectrum**



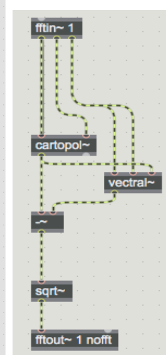

**Analyze Spectrum**



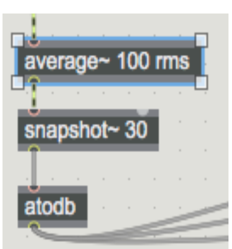

Centroid




Flatness




Flux




Volume

## Reference List

Anderson, J. (2000). *A provisional history of spectral music*. Contemporary Music Review 19(2), 7–22.

Arrell, C. (2003). *The music of sound: An analysis of partiels by Gérard Grisey.* In Spectral World Musics: Proceedings of the Istanbul Spectral Music Conference, edited by Robert Reigle and Paul Whitehead. Istanbul, Turkey: Pan Yayincilik.

Anderson, J. (2001) Sadie, S. (ed)., Tyrrell, J. (ed). Spectral music. In *The New Grove dictionary of music and musicians* second edition, London, England: Macmillan Publishers.

Beauchamp, J. W. (1971). [Review of *An Introductory Catalogue of Computer Synthesized Sounds*]. *Perspectives of New Music*, *9/10*, 348–350. http://doi.org/10.2307/832154

Bennett, B. *The principles of spectralism and Tristan Murail gondwana.* Retrieved on April 12, 2016

Braddell, R. (2000). *Stimmung for 6 vocalists.* Retrieved May 30, 2016, from http://homepage.eircom.net/~braddellr/stock/

Brown, M. (2007). *Tearing down the wall of sound: The rise and fall of Phil Spector.* New York: Alfred A. Knopf.

Burk, P., Polansky, L., Repetto, D., Roberts, M. & Rockmore, D. *Music and computers: A theoretical and historical approach. Retrieved from* music.columbia.edu/cmc/musicand*computers*

Burton, A. (2006). *Kaija Saariaho Biography.* Retrieved on April 23, 2016 from http://saariaho.org/wp-content/uploads/2015/09/150723-Bio-ENG-KS.pdf

Charles, J.F. (2009). *A tutorial on spectral sound processing using max/msp and jitter.* Computer Music Journal, 32(3), 87-102

Chowning, J. (1973). *The synthesis of complex audio spectra by means of frequency modulation.* Journal of the Audio Engineering Society, 21(7), 526-534

Christie, D. (1998). Merzbow's disciple, decibels and deity: Japan's minister of sonic terror turns on the feedback. Digi-zone Online Entertainment. Retrieved on January 25, 2017 from https://web.archive.org/web/19980201190109/http://www.digi-zine.com/17merz.htm

Clarck, W. (2009). Unstable Molecule. Wired Magazine, 308, 32-37. Retrieved June 6, 2016, from http://www.spectralmusic.org/Iancudumitrescu/Thewire.html

Cont. A. (2012). *Voice to music score transcription.* Retrieved April 5, 2016 from http://forumnet.ircam.fr/tribune/making-an-orchestra-speak

Cook, P. R. (2002). *Real sound synthesis for interactive applications*. Natick, Mass: A K Peters.

Corniclło, A. (2000). *Timbral organization in Tristan Murail's desintegrations and rituals (doctoral dissertation) Brandeis University, Massachusetts Pixel Formats.* (n.d.). In Derivative. Retrieved Febraury 29, 2016, from https://www.derivative.ca/wiki088/index.php?title=Pixel_Formats

Dodge, C., & Jerse, T.A. (1997). *Computer music: Synthesis, composition, and performance.* New York: Schirmer Books.

Dudas, R. *Max/MSP Tutorial 26.* Cycling 74

Dufourt, H. (1985-1986) *Musique spectrale.*Consequences, 7, 114-115

Dumitrescu, I. (1996-1997). *Remote pulsar.* Retrieved February 13, 2014 from http://spectralmusic.org/Iancudumitrescu/

Fineberg, J. (2000a). *Spectral music: History and techniques*. Netherlands: Overseas Publishers Association.

Fineberg, J. (ed.). (2000b). *Spectral music: Aesthetics and music.* Amsterdam, Netherlands: Overseas Publishers Association.

Fineberg, J. (2000c). *Guide to the basic concepts and techniques of spectral music.* Contemporary Music Review 19(2)

Fineberg, J., Daubresse, E., & Assayag, G.(2000d). Technology and creation: The creative evolution. Contemporary Music Review, 19(2), 61-80.

Fineberg, J. (2012). *Grisaille.* France: Gérard Billaudot Éditeur.

Fineberg, J. (2014). *Biography.* Retrieved on June 1, 2016 from https://joshuafineberg.com/biography/

Frymier, P.D., Ford, R.M., Berg, H.C. & Cummings, P.T. (1995). *Three-dimensional tracking of motile bacteria near a solid planar surface*. PNAS, 92(13): 6195-6199.

Gray, A., & Markel, J. (June 01, 1974). A spectral-flatness measure for studying the autocorrelation method of linear prediction of speech analysis. *Transactions on Acoustics, Speech, and Signal Processing, 22,* 3, 207-217.

Grey, J. M. (1978). Perceptual effects of spectral modifications on musical timbres. *The Journal of the Acoustical Society of America, 63 (*5), 1493-1500.

Grisey, G. (1980). *Partiels,* Paris: Sté Ame des Editions Ricordi

Grisey, G. (2000). *Did you says spectral?,* Translated by Joshua Fineberg. Contemporary Music Review, 19(3), 1-3

Hardesty, L. (2009), *Explained: The discrete fourier transform*, Retrieved from http://news.mit.edu/2009/explained-fourier

Harvey, J. (2000). *Spectralism*. Contemporary Music Review, 19(3), 11-14Harevy, J. (1996). *Trombeau de messiaen.* London, England: Faber Music Ltd.

Harvey, J. (1975) *Inner Light III.* Retrieved on June 10, 2016 from http://brahms.ircam.fr/works/work/9027/

Holz, K. (2013). *The Acoustics of the clarinet: An observation of harmonics, frequencies, phases complex acoustic impedance and resonance.* Abilene, TX: Abilene Christian University.

Hodgikinson, T. (Interviewer) & Dumitrescu, I. (Interviewee). (1997) Resonance, Volume 6 [Interviewer Transcript] Retrieved from Iancu Dumitrescu's website: http://www.spectralmusic.org/Iancudumitrescu/Hodgkinson.html

Honegger, A. (1923). *Pacific 231. Mourice Senart.* Chateau, Paris: Maurice Senart.

Hurel, P. (2007). *Spectral music: Long-term perspectives.* Retrieved on March 14, 2016 from http://www.philippe-hurel.fr/en/musique_spectrale.html

Krapp, P. (2011). *Noise channels: Glitch and error in digital culture*. Minneapolis, MN: University of Minnesota Press.

Klingbeil, M. (2009). *Spectral analysis, editing and resynthesis: Methods and applications.* New York, NY: Columbia University.

Kubitschek, B. (2016). *Our cosmic address.* Retrieved from www.nerdsinc.com/blog-1/2016/3/14/challenge

Lechner, P. (2014). *Multimedia programming using Max/MSO and Touch Designer.* Birmingham: Packt Publishing.

Lehman, S. (n.d.). *Steve Lehman*. Retrieved April 24, 2015, from http://www.stevelehman.com/bio

Lehman, S. (2009). *Travail, Transformation and Flow*. Recorded by Steve Lehman Octet. Brooklyn, NY: Pi Recording.

Lehman, S. (2012). *Liminality as a framework for composition: Rhythmic thresholds, spectral harmonies and afrological improvisation.* Dissertation. New York, NY: Columbia University.

Long, S. *Basics of the spectrum analyzer.* Retrieved January 21, 2016, from ece.ucsb.edu/~long/ece145a/spec_analyzer.pdf

Lovely, P., & Dahlquist, F. (1975). Statistical measures of bacterial motility and chemotaxis. Journal of Theoretical Biology, 50: 477-496.

Mao-Takacs, C. (2014). *A conversation with Kaija Saariaho*. Retrieved April 23, 2015, from http://www.musicandliterature.org/features/2014/9/22/a-conversation-with-kaija-saariaho

Mabury, B. (2006). *An investigation into the spectral music idiom and its association with visual imagery, particularly that of film and video*. Perth: Australia: West Australian Academy of Performing Arts.

McClellan, J. H., Schafer, R. W., & Yoder, M. A. (1998). *DSP first: A multimedia approach*. Upper Saddle River, NJ: Prentice Hall.

MacMillan, K. (2015). *Biography: Kaija Saariaho.* Retrieved on June 6, 2016 from http://saariaho.org/biography/

Moscovich, V. (1997). *French spectral music: and introduction.* Tempo, 200, 21-27

Nam, U. (2001). *Special are exam part II.* Retrieved January 27, 2016 from https://ccrma.stanford.edu/~unjung/AIR/areaExam.pdf

Niinisalo, J. (2010). *The aphex face.* Retrieved April 3, 2016 from http://www.bastwood.com/?page_id=10

Noetinger, J. (Interviewer) & Dumitrescu, I.  (Interviewee). (1996-1997). Revue et Corrigée Volume: 29, Genoble & Bananafish Volume: 12, San Francisco [Interview Transcript] Retrieved from Iancu Dumitrescu's website: http://spectralmusic.org/Iancudumitrescu/Noetinger.html ordinal number. (2011). In *Merriam-Webster.com.* Retrieved February 2, 2017, from https://www.merriam-webster.com/dictionary/ordinal%20number?utm_campaign=sd&utm_medium=serp&utm_source=jsonld

Park, T. H. (2010). *Introduction to digital signal processing: Computer musically speaking*. Singapore: World Scientific.

Parkinson Lab. (2003). An overview of E. Coli chemotaxis. Retrieved March 4, 2016 from http://chemotaxis.biology.utah.edu/Parkinson_Lab/projects/ecolichemotaxis/ecolichemotaxis.html

Peeters, G. (2004). *A large set of audio features for sound description in the CUIDADO project.* Retrieved February 1, 2016 from http://recherche.ircam.fr/equipes/analyse-synthese/peeters/ARTICLES/Peeters_2003_cuidadoaudiofeatures.pdf

Peeters, G., Giordano, B. L., Susini, P., Misdariis, N., & McAdams, S. (2011). The Timbre Toolbox: extracting audio descriptors from musical signals. *The Journal of the Acoustical Society of America, 130,* 5, 2902-16.

Pousset, D. (2000). *Spectralism*. Contemporary Music Review, 19(3), 67-110

Randall, J. K., Vercoe, B., Dodge, C., Kessler, M., Columbia University., & Princeton University. (1970). *Computer music*. New York, NY: Nonesuch Records.

Reigle, R. (2008). *Spectral musics old and new.* In Spectral World Musics: Proceedings of the Istanbul Spectral Music Conference, edited by Robert Reigle and Paul Whitehead. Istanbul, Turkey: Pan Yayincilik.

Risset, J-C. (1969). *An introductory catalog of computer synthesized sounds.* Murray

Hill. New Jersey: Bell Laboratories.

Roads, C., & Strawn, J. (1995). *The computer music tutorial*. Cambridge, MA: MIT Press.

Rose, F. (1996). *Introduction to the pitch organization of French spectral music.* Perspectives of New Music, 34(2), 6-39

Saariaho, K. (1990). *Oi Kuu.* Copenhagen, Denmark: Wilhelm Hansen Copenhagen.

Sargenti, S. (2013). *Techniques and technologies in the composition Mantra: Some analytical considerations.* Journal of Literature and Art Studies, 3(9), 533-567

Service, T. (2012). *A guide to Kaija Saariaho's music.* Retrieved on May 30, 2016 from https://www.theguardian.com/music/tomserviceblog/2012/jul/09/kaija-saariaho-contemporary-music-guide

Ungvary, T., Waters, S. (1992). *The sonogram: A tool for the documentation of musical structure.* Ex-Tempore, 6(1)

Winckel, F., & Binkley, T. (1967). *Music, sound and sensation: A modern exposition*. New York, NY: Dover Publications.

World Soundscape Project., & Truax, B. (1978). *The World Soundscape Project's Handbook for acoustic ecology*. Vancouver, B.C: A.R.C. Publications.

Wood, S. (2005). *Beginners guide to Praat*. Retrieved January 21, 2016, from swphonetics.com/praat/tutorials/what-are-formants/