Environmental Sonic Translation

Christopher Luna-Mega
Mexico City, Mexico

B.A., Universidad Iberoamericana, 2002
B.M., Escuela Nacional de Música, UNAM, 2010
M.A., Mills College, 2013
M.A., University of Virginia, 2018

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Matthew Burtner, Ph.D.
Ted Coffey, Ph.D.
Leah Reid, D.M.A.
Judith Shatin, Ph.D.
Willis Jenkins, Ph.D.

Abstract

Environmental Sonic Translation offers a collection of compositional techniques drawing on detailed analyses of environment-derived soundscapes, which can be found in geological, biological, and anthropogenic sources. It relies on audio technology to collect sounds, analyze their spectra and reassign their features to musical instruments. This research is situated within the fields of Ecoacoustics/Acoustic Ecology, Spectralism, and *musique concrète*, building a framework that responds to the aesthetic, conceptual and technical principles of *environmental sonic translation*. The main purpose of this work is to embody the various ways in which sound exists in the world.

In this dissertation, I offer a systematic analytic and notational methodology applicable to a wide field of soundscapes, culminating in a compositional body of work generated with the techniques presented. As a background, I discuss influential work that has contributed to the field of environmental sonic translation.

Acknowledgements

This dissertation is the culmination of an epiphany that I had more than a decade ago while listening to the sound of a stream in the Lagunas de Zempoala National Park, about 40 miles South-West of Mexico City, my hometown. I was absorbed in a deep state of perception, listening to the patterns of subtle pitches, dynamic shifts, rhythmic shapes, and continuous flow of the stream. As in other moments of connection with nature in which I had the desire of being a tree or a bird, I imagined being the stream. The thought came to me that I must orchestrate that stream in a way that the orchestra sounds just like it—becomes it—, featuring every aspect of its noise and its pitch, of its energy and complexity. I had no idea how I was going to do this, when my tools were limited to my ears, notation, orchestration, and composition skills. Fourteen years later, I have not been able to reproduce the sound of the stream, of nature, but I have been able to embody it, to know its acoustic structures in ways that I did not suspect that afternoon when I had the first dream of this compositional endeavor.

This would have not been possible without the help of my advisors, colleagues, family and friends. Thanks to my advisors Matthew Burtner, who has been a model of enthusiasm, creativity and possibilities in the academic field; Judith Shatin, whose insights on idea, presentation, form, and composition technique led me to a new stage of maturity; Leah Reid, for her invaluable technical, methodological, and literature resources for timbre-based composition; Ted Coffey, whose artistic sensibility and conceptual refinement have nourished my compositional search; and Willis Jenkins, professor of Religious, Ethics, & Environment, for serving as an outside reader and being an inspiration and example for interdisciplinary and collaborative work. Thanks to my generous and bright collaborators whose contributions were crucial to a portion of the pieces included in this dissertation: Joshua Jones (Scripps Institution of Oceanography), Jon Gomez (UVa, Data Science Institute), and Max Tfirn (Christopher Newport University, Department of Music). Thanks to Travis Thatcher, whose technical support and advice were essential for the performances and recording sessions of the music here presented. Thanks to the Jefferson Scholars Foundation, whose fellowship offered me the material resources and a diverse intellectual community that has provided insightful feedback to this dissertation's contents.

Special thanks to my wife, Virginia Luna-Mega, for knowing every spiritual connection to every sound composed for this dissertation, as well as for her moral support throughout the challenges and joys of the doctoral odyssey; to my son, Santiago, for infusing me with energy and inspiration to give my best focus to this work; to my parents, Joy Mega-Simmons and Sergio Luna-Sefami, for having given me the childhood seeds and soil for music, art, exploration, friendship, and nature; and to my sister Lorena Luna-Mega, whose strength, love, and lightness have helped me to step solidly throughout my path. I also thank my uncle and aunt, Mark and Lourdes Mega, for providing me with the vision of an inspiring future as an academic since I was a child—this fueled my will to pursue this path. I wish to express my deep gratitude to Bernardo and Lupita Ávalos for endorsing the foundational stages of my work as a composer, without which completing my musical education in Mexico City would have not been possible. Thank you to Elias McQuade and Bogdan Krstic for their sincere friendship and for having joined me in the tennis courts, soccer fields and hiking trails, restoring my mental and physical health throughout for the last six years. And finally, many thanks to Aaron Stepp, Alex Christie, Ben Luca Robertson, Eli Stine, Kevin Davis, Omar Fraire, Tim Booth, and the Composition and Computer Technologies PhD students during my time at the University of Virginia. More than colleagues with whom I share ideas they are teachers from whom I learn invaluable lessons of music and life.

Contents

Introduction

Environmental sonic translation refers to music that has been specifically derived from acoustic features of environmental sound. It results in translation, transcription and orchestration of any given sound, whether biological, geological or anthropogenic. This work relies on recording technology and the resulting acoustic representations of sound, such as temporal representations—emphasizing the amplitude of a sound over time—and spectral representations—classifying the range of different shapes that waves can take, modelling sound through the superimposition of any number of waves of different frequencies with each individual wave taking the form of a "sinusoid." (McAdams, Depalle, and Clarke, 2004) Through the use of such methods, sound models are extracted from environmental soundscapes. The models function as a maquette for the resulting music, which shares acoustic and syntactical features with the original source. The ultimate purpose of this environmental sonic translation is to enrich our understanding of natural and anthropogenic phenomena and to use it as a foundation for music composition.

Strongly connected to the field of ecoacoustics, this approach shares its aim "to establish an innovative biosemiotic narrative to ensure a strategic connection between nature and humanity." (Farina, 2018, p. 21) From a perspective of Acoustic Ecology, it seeks the perceptual and performative embodiment of environmental sound, drawing from the three kinds of environments that exist in the physical world: *geophony*, *biophony* and *anthropophony* (Krause, 2012). From the spectral perspective, this framework uses recording technology, music notation and orchestration to collect such sounds, analyze their acoustic spectra, and reassign them to musical instruments. From a soundscape classification perspective, Lasse Thoresen's broad categories of pitched and complex soundscapes—an adaptation of *musique concréte* typomorphologies developed by Pierre Schaeffer—are used to situate the body of work derived from this research (Thoresen, 2004).

The methodologies that I have used for the four selected compositions detailed here, as well as the findings resulting from their analyses, form the core of the dissertation. As a background, I offer an overview of the work that has preceded and influenced the compositional concepts here presented. This overview includes historical examples, concepts of acoustic ecology, spectral techniques, and analyses of work that have inspired and influenced this research. A selection of the original compositions for various ensembles—from which the methodologies and analyses for this dissertation are drawn—are included in the appendix. These compositions are included in my album *Aural Shores*, Environment-derived composition, Vol. 1.

Chapter 1 summarizes the relevant and most recent historical events that precede my research and compositional work. These are: 1) Russolo and Cage; 2) *musique concrète* 3) Messiaen; 4) Acoustic Ecology; and 5) the Spectral school.

Chapter 2 presents compositional, analytical and computer-assisted composition tools employed in environmental sonic translation.

Chapter 3 features two analyses of work by composers that have written influential sound model-based compositions: 1) *Korwar*, by François-Bernard Mâche; 2) *Quadraturen IV*, by Peter Ablinger.

Chapter 4 features the analytical, compositional and orchestration techniques that I have derived from different types of sound models. The chapter presents a selection of the compositions written under the light of the concepts explored during the process of writing the dissertation. These pieces are organized based on their compositional and analytical responses to Thoresen's aforementioned two broad categories of sounds: complex sounds and pitched sounds. *Geysir,* for seven piano parts and electronics, and *Waves Break Aural Shores*, for saxophone quartet and electronics—derived from geyser and ocean recordings, respectively—feature the techniques focused on noise. *Night Music,* for reed quintet and electronics, and *Under the Sea Ice*, for string quartet and electronics—derived from deciduous forest insects and Arctic bearded seals recordings, respectively—feature techniques focused on pitched soundscapes.

The appendix includes the body of environmental sonic translations that led to and was derived from this dissertation's research. It also includes *Aural Shores*, an album released in April, 2020, which includes the four pieces detailed in Chapter 4.

I. Historical Background of Environmental Sonic Translation

The scientific approach to generating and translating a sound model, which derived from the invention of recording technology and computer analysis of sound, elicits an important distinction between environmental sonic translation and a substantial amount of music from various periods of Western and non-Western cultures that is inspired by and translates sound models of nature and human activity, including program music and ritual music. The main difference lies in the degree of connection of the features or data[1] of the original sound to the various aspects that constitute the musical work. Both involve translations. However, the former seeks to base compositional and musical choices from quantifiable acoustic components of the sound itself. In environmental sonic translation, an essential objective is, in the words of composer and musicologist François-Bernard Mâche, "to try to bring together poetics and theory, and to show the advantages that there are in advancing an aesthetic project on the basis of a harmony with natural data." (Mâche, 1983, 166) The aims of this statement consist of using values of musical parameters found in nature in the poetics of music creation. The inclusion of a scientific perspective for listening to the world in the compositional process is a core feature of this sound model-based instrumental composition.

Composer Fabio Cifariello-Ciardi provides a useful distinction towards the definition of the field of environmental sonic translation, where "the sound and the network of relationships that define it are not simply chosen in order to be used within a predefined syntactic framework, but instead they are considered as elements that can cause profound changes in the strategies and syntaxes used by the composer." (Cifariello-Ciardi, 2008, 124) In this conception of compositional content and syntax, the physical components of the sound itself, detailed in the sound-model, become the foundation of the musical materials and/or the form itself of a musical composition.

While such distinctions between the compositional approach here proposed and its numerous predecessors are essential to understand environmental sonic translation, this approach has undoubtedly been shaped aesthetically, technically and technologically from specific musical practices and contemporary scientific approaches. Since environmental sonic translation follows

[1] According to the Cambridge Dictionary, "information, especially facts or numbers, collected to be examined and considered and used to help with making decisions."

the continuum of an inextinguishable amount of predecessors from several centuries of both Western and traditional world music that would require several chapters to discuss, this background-introductory chapter focuses solely on its most recent and direct influences.

1.1. Embracing Noise: Luigi Russolo and John Cage

In 1913, futurist Luigi Russolo, in *The art of noise*, introduced the idea of systematizing the musicality of the various noises of the world, mainly the anthropogenic noises introduced by the industrial revolution. This established the compositional idea that each noise has a harmonic signature. Half a century later, when the Fast Fourier Transform allowed detection of the partials and amplitudes integrating noise, the following idea of Russolo's was explored in depth by some environmental sonic translation practices.

> We want to score and regulate harmonically and rhythmically these most varied noises. Not that we want to destroy the movements and irregular vibrations (of tempo and intensity) of these noises! We wish simply to fix the degree or pitch of the predominant vibration, as noise differs from other sound in its irregular and confuse vibrations (in terms of tempo and intensity).
>
> Each noise possesses a pitch, at times even a chord dominating over the whole of these irregular vibrations. The existence of this pre-dominant pitch offers us the technical means of scoring these noises, that is to say to give to a noise a certain variety of pitches without losing the timbre that characterizes and distinguishes it. Certain noises obtained through a rotating movement can give us a complete ascending or descending scale through the speeding up or slowing down of the movement. (Russolo, 1913, p. 9)

Twenty-five years later in 1938, John Cage was also imagining the possibilities of using the environment as the main musical material in the same decade when magnetic tape was invented. In *The Future of Music: Credo*, he states the following:

> Wherever we are, what we hear is mostly noise. When we ignore it, it disturbs us. When we listen to it, we find it fascinating. The sound of a truck at fifty miles per hour. Static between the stations. Rain. We want to capture and control these sounds, to use them not

> as sound effects but as musical instruments (...) Whereas in the past, the point of disagreement has been between dissonance and consonance, it will be, in the immediate future, between noise and so-called musical sound. (Cage, 1961, p. 3)

1.2. *Musique Concrète*: Recorded Sound as Material for Composition

A professional engineer born of musician parents, Pierre Schaeffer began experimentation with recorded sounds after he joined RTF (Radiodiffusion-Télévision Française) in 1936. In 1951, he created the Groupe de Recherche de Musique Concrète (GRMC, now GRM), in RTF. With this initiative he established a studio with a magnetic tape recorder. Schaeffer thought of "the 'materialization' of the sound in the form of a recording. In these experiments, 'sound was apparently no longer evanescent, and it kept its distance from its cause: it acquired stability; we could manipulate it, copy it, vary its energetic dimensions, without being bound by its initial contingencies.'" (Schaeffer, 1966, p. 51) This concept became the foundation of *musique concrète*. In this new method of composing, recorded sounds of musical instruments and various objects, natural and anthropogenic environments, the human voice and electronically processed sounds were used as raw material and assembled as a *montage*. Environmental recordings are an important element in the work of Schaeffer and the composers associated to *musique concrète* and the GRM. Schaeffer's *Etude aux Chemins de Fer* (Study of Locomotives) uses recordings from the Gare des Batignolles station in Paris. This piece embodies Russolo's notions of noise as musical material. Pierre Henry's *Variations for a Door and a Sigh* features sounds of a creaking door, a musical saw and human breath. Luc Ferrari's *Presque Rien no. 1* consists of a recording of the sounds of a town by the sea. In *De Natura Sonorum: Matiéres Induites*, Bernard Parmegiani uses the sound of flowing water and blends it with percussive objects with similar sonic patterns.

In 1966, Schaeffer wrote his *Traité des objets musicaux*, in which he provided a detailed classification of sounds based on their spectral features, durations, and behavior. This classification derived in what now is called *typomorphology*. In the Twenty-First Century, Lasse Thoresen developed Schaeffer's classification in the direction of a practical tool for conceptualizing and notating sound quality. In his *Spectromorphological analysis of sound objects,* Thoresen establishes two broad categories for analysis: Pitched and Complex sounds. (Thoresen, 2004) These categories are the basis for the structure of Chapter IV of this dissertation, in which the composed work and its methodologies are organized as either Pitched or Complex soundscapes.

1.3. Olivier Messiaen

Olivier Messiaen was one of the first composers to use recording technology as a resource for translation of the environmental sound model into music notation. His catalogue includes a significant number of pieces derived from birdsong models that were notated –mostly traditionally and determinately– either in situ or while listening to recordings. "The cahiers' (notebooks) marginalia also incorporate comments on timbre, musical mood, tempo, and proposed orchestration, as well as Messiaen's personal reaction to the natural world." (Taylor, 2014, p. 71) Before the 1960's, Messiaen used only his ears, his memory, manuscript paper and pencil for his transcriptions. *Le merle noir* (1952), for flute and piano, was the first piece completely based on birdsong, specifically the song of the blackbird. In *Réveil des oiseaux* (1953), for orchestra, Messiaen orchestrates several transcriptions of birdsong that take place between midnight and noon in the Jura mountains, in the Western Alps. *Catalogue d'oiseaux* (1956-1958), for piano, is a 7-book comprehensive work with piano arrangements of transcriptions of various species of birds. In the 1960's, Messiaen began to use a tape recorder to record birds for further reference.

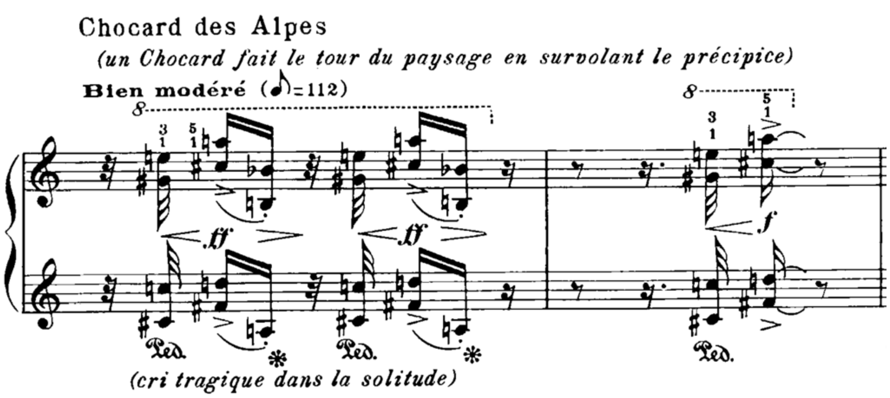


Figure 1. Messiaen, *Catalogue des Oiseaux,* Book 1, p. 6

1.4. Acoustic Ecology: R. Murray Schafer, Barry Truax, Bernie Krause

Two decades after Pierre Schaeffer founded the GRM and treated recorded sounds as objects for *musique concrète* compositions, composers and researchers in the Americas turned their attention from the specific individual sounds to the totality of sounds in various environments.

R. Murray Schafer, a musician, composer, and pedagogue at Simon Fraser University (SFU), founded the World Soundscape Project (WSP) as an initiative to draw attention on the increasing noise pollution in Vancouver. The WSP included his students—Hildegard Westerkamp, Barry Truax, Howard Broomfield, Peter Huse and Bruce Davis—who were composers and activists committed to recording the soundscapes in various locations and giving lectures and workshops. (Simon Fraser University, 2020) Schafer suggests that we try to hear the acoustic environment as a musical composition and further, that we can be "simultaneously its audience, its performers and its composers." (Schafer, 1977, p. 205) In a similar vein, Truax defined *natural sound composition* as a recorded soundscape whose organization takes attention for its variety, textures, timbres, and other acoustic parameters, and whose recording can be listened with the same appreciation as one listens to music. (Truax, 2001)

Bernie Krause is a musician and soundscape ecologist, founder of Wild Sanctuary, an organization dedicated to archiving natural soundscapes. He proposes to listen to the collective sound expressed by an entire biome, or habitat. By analyzing the sonic density and texture of natural soundscapes, Krause has been able to detect declining biomes that would be more difficult to identify through images and other methods of observation. (Krause, 2012) His *Niche hypothesis* states that "each creature appears to have its own sonic niche (channel, or space) in the frequency spectrum and/or time slot occupied by no other at that particular moment. (…) An audio bio-spectrum is an acoustical spectrographic mapping of any particular habitat by frequency (pitch, sometimes tone) and amplitude (loudness) over short periods of time." (Krause, 1993, p. 2-3) *Niche hypothesis*-based sonogram analyses were implemented in the pre-compositional work for *Night Music*, for reed ensemble and electronics, discussed in Chapter 4.2.2.

Taking Schafer's and Truax's aforementioned ideas, Mexican composer Manuel Rocha Iturbide developed structural analyses that are important antecedents to the environmental sonic translation methodologies presented in this dissertation. He mixed graphic acoustic information with some basic music notation of fixed recordings of environmental soundscapes he collected in

Chichicastenango, Guatemala; Hoi An, Vietnam; Río Caura, Venezuela; and Mexico City, Mexico. In Rocha's words, this work "can teach us in a more or less objective way the different sonic structures that exist in the world around us, and that it could suggest to us as composers, structures that we could then imitate or develop in electroacoustic music or instrumental composition." (Rocha, 2009, p. 459)

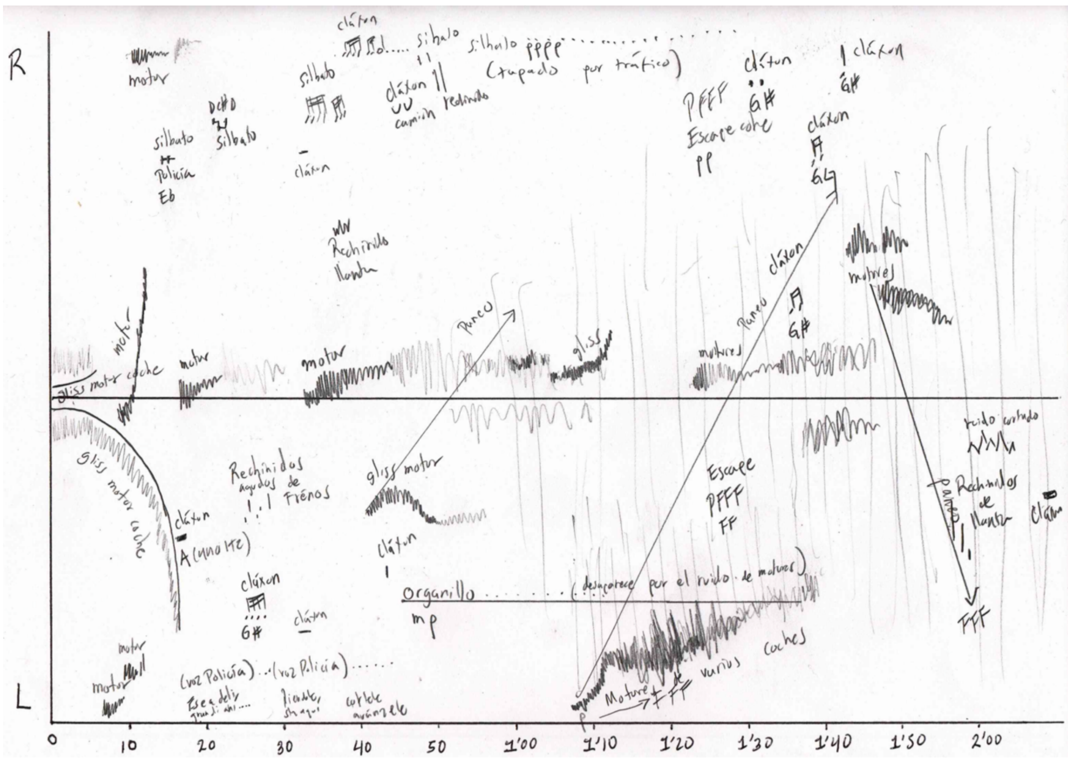


Figure 2. Manuel Rocha, transcription of a soundscape in Eje Central Ave., Mexico City

1.5. The Spectral School

Composers associated with or technically related to the spectral movement from the mid-1970's to the present day, including Per Norgaard, Tristan Murail, Gerard Grisey, Jonathan Harvey, Claude Vivier, Mesias Maiguashca, Peter Eotvos, Philipe Hurel, Marc-Andre Dalbavie, Kaija Saariaho, Joshua Fineberg, Horatiu Radulescu and Iancu Dumitrescu, explored sound models of acoustic spectra through various processes. Their instrumental works were based on spectral analyses and processes of musical instruments, the human voice and percussive objects, as well as the abstract model of the harmonic series. Based on the work of Fourier, who demonstrated

Maiguashca amongst them; several other composers, notably Claude Vivier and Peter Eotvos, appear to have maintained occasional contact with the group." (Anderson, 2000, p. 15) Later in the nineties, *L'Esprit des Dunes* (1994) develops melodic events from various spectra. Since the foundation of *L'Itinéraire* to the present day, composers who do not identify as spectralists nevertheless utilize various spectral approaches (i.e., timbre analysis) to compose their sound models. This has resulted in an expansion of the ramifications of the spectral approach.

Due to such breadth of the spectral approach to composition, as well as the fundamental interest of spectral composers in the structure of sound, the spectral technique is arguably the most prominent in the field of environmental sonic translation into instrumental composition. Pressnitzer and McAdams maintain the following:

> The spectral intuition consisted in founding compositional systems on the structure of sound, and thus in deriving fields of musical relations from sound itself. The wager of such an approach is to give to a listener reference points that are naturally understandable, while allowing the use of the new potential offered by micro-compositional work at the level of sound. (Pressnitzer & McAdams, 2000, p. 34)

François-Xavier Féron shows in more detail how in Grisey's *Dérives*, *Periodes*, and *Partiels*, the spectrum "not only represents a pool of singular pitches, but also acts as the source of numerical values that both afford control over a large number of compositional actions, in addition to the temporal structure of the piece." (Féron, 2011, 349) This became the foundation of numerous sound-model-based instrumental compositions, such as Harvey's *Mortuos Vivo, Plangos Voco* (1980), Saariaho's *Verblendungen* (1984), and Ablinger's *Quadraturen IV* (1998).

The influence and scope of these techniques is so expansive that "there is no real school of spectral composers; rather, certain fundamental problems associated with the state of contemporary music, since at least 1965, have repeatedly provoked composers from widely different backgrounds into searching out some common solutions involving the application of acoustics and psycho-acoustics to composition." (Anderson, 2000, p. 7)

1.5.2. Psychoacoustics

In “A Provisional History of Spectral Music”, Julian Anderson states that the most superficial feature of spectral music is the use of harmonic spectra as the basis for composition. Rather, “psychoacoustics is a more fundamental concern.” (Anderson 2000, p. 8) Psychoacoustics plays an important role in the analysis and composition phases of environmental sonic translation because, due to the fact that a considerable amount of sounds in a signal may contain various sources, “the brain has to ‘decide’ which bits of sound belong together, and which bits do not. The grouping of sounds into perceptual units (events, streams, and textures) determines the perceived properties or attributes of these units.” (McAdams et al., 2004, p. 183) The following auditory representations play an essential role in the analysis of sound models:

a) vertical organization (perceptual fusion): groups “what is likely to come from the same acoustic source, and (separates) it from what is coming from different acoustic sources.” Grouping cues: 1. Spatial source; 2. Vibration system source (timbre) (Pressnitzer & McAdams, 2000, p. 50)
b) horizontal organization (streams): in the realm of temporal evolution, it “studies the formation of auditory streams. A stream is a sequence of events that can be considered to come from a single source. (…)” (Pressnitzer & McAdams, 2000, p. 50) Quoting the work of McAdams and Bregman on hearing musical streams (1979), Pressnitzer and McAdams state that “the general law governing the formation of streams appears to be based on spectral continuity.” (2000, p. 50) Grouping cues: 1. Frequency, 2. Time proximity (synchrony) and 3. Timbral similarity/harmonicity

Timbre, which has been mentioned in the auditory representation models above, is an essential concept in psychoacoustics. “The ‘official’ scientific definition (of timbre) is a non-definition: the attribute of auditory sensation that distinguishes two sounds that are otherwise equal in terms of pitch, duration, and loudness and that are presented under similar conditions.” (McAdams et al., 2004, p. 190) A more precise definition of timbre provides attributes in terms of which timbres may be distinguished: Spectral centroid; Attack quality; Smoothness of the spectral envelope; Evolution of the envelope; Roughness; Noisiness/inharmonicity (2004). These attributes play an important role in the perceived fusion and streams of events in environmental soundscapes.

The organizational principles of psychoacoustics discussed above may be used as compositional tools. In *Désintégrations* (1982), Murail presents a melody with a set of intervals in rapid tempo. This complex melody is perceived as fused (Pressnitzer & McAdams, 2000, p. 53)—defined as *intended fusion* by McAdams, et. al. In a different section of the piece, that melody is heard 'disintegrated' in a section of stasis. In my piece, *Waves Break Aural Shores*, there is a juxtaposition between intended fusion and separation of the ocean's partials through shifting the time proximity of the saxophones' onsets, the frequency range of the material and the timbral similarity/difference of their registers. In sections of stasis in the frequency domain the saxophone quartet is perceived as a single stream, while in sections of saturated melodic material and separation in register, the material is perceived as separate streams.

Figure 3. Murail, *Désintégrations* (1982), perceptual fusion (in section VII)

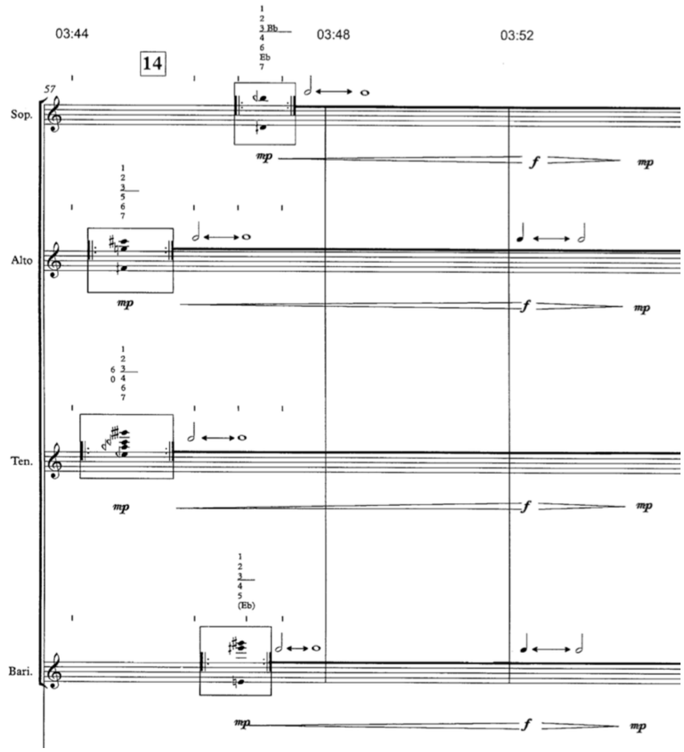


Figure 4a. *Waves Break Aural Shores*, intended fusion through timbral and register similarity

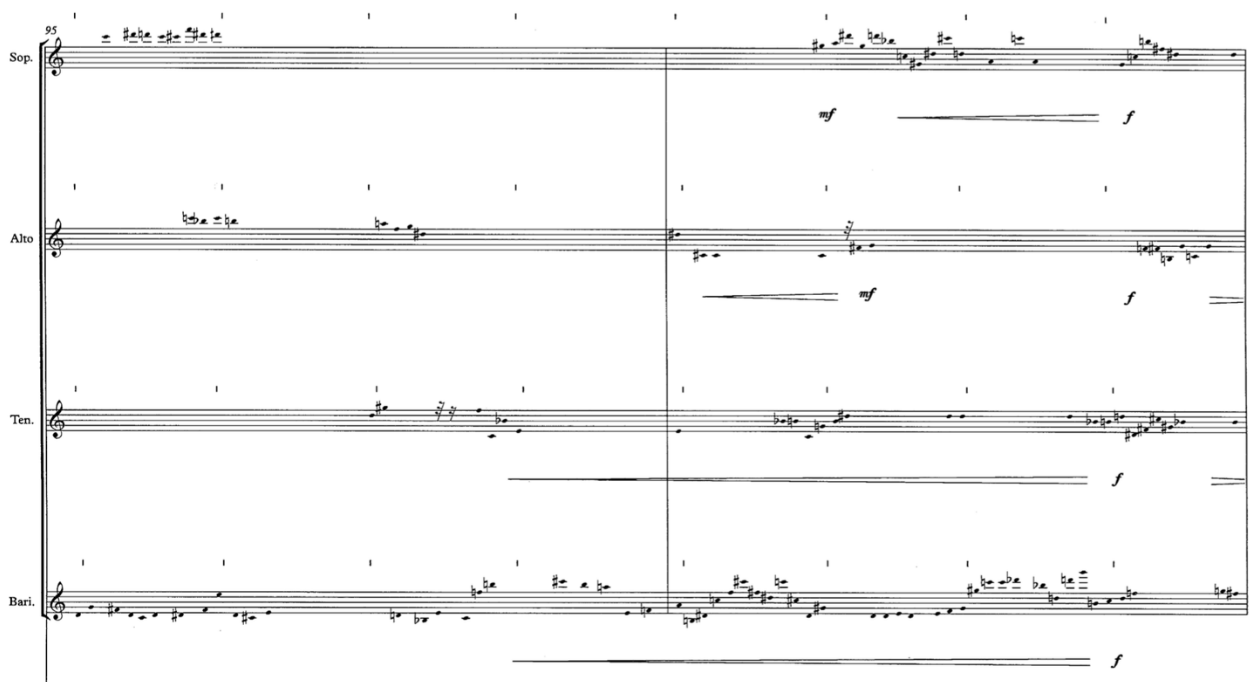


Figure 4b. *Waves Break Aural Shores*, separation through distance in register/timbre and varying onsets.

1.5.3. Instrumental additive synthesis

Spectral composers assign the partials of the analyzed sound to specific pitches or groups of pitches performed by an instrument or various instruments, a technique that is oftentimes referred to as "instrumental additive synthesis." While the model presents a fundamental frequency and its sine tones or partials, the translation of the model to instrumental music converts its sine tones into fundamental frequencies with a different set of partials. In this sense, François Rose observes that "the idea is not to create an acoustical reproduction of an electronic sound, but rather to adapt an electronic procedure for acoustical instruments. Naturally, the result of this procedure, while deriving from physical models, no longer shares but replaces the characteristics of the modeled phenomenon." (Rose, 1996, p. 11) For this reason, environmental sonic translation is not concerned with an instrumental mimesis of the environment. Instead, it is concerned with the *translation* of the environment's sonic features into the musical and instrumental realm.

1.5.4. Harmonicity / Inharmonicity

An important feature in spectral music is the flow between harmonicity and inharmonicity as a strategy of contrast and development.

> Timbre manipulation opens up the possibility to look for a continuous scale that could reproduce, in some respects, the expressive means associated with the tonal notions of consonance and dissonance. The exploration of such a dimension has been undertaken by many composers, essentially in an intuitive fashion. Tristan Murail, for example, has ordered timbres and aggregates with a measure of inharmonicity (*Desintégrations*, 1982). Kaija Saariaho has defined a sound/noise axis intended to reproduce the harmonic capacity to create tension and relaxation (*Verblendungen*, 1982-1984). Joshua Fineberg has adopted a hierarchy founded on the pitch of virtual fundamentals (*Streamlines*, 1995). (Pressnitzer & McAdams, 2000, p. 41)

Different patterns of flow between harmonicity and inharmonicity are present in each of the original environmental sonic translations in this dissertation. For example, both the instrumental and field recording components in *Night Music* present a clear flow from harmonicity to inharmonicity (Figure 5). The clear and thin horizontal lines in the beginning show sounds from the instruments and the field recordings with higher degrees of harmonicity. Broader bandwidth in the lines towards the end, as well as an overall prominence of sound (across all the frequency spectrum) reveal a higher presence of inharmonicity (noise). This is a signature of the summer dusk soundscape in deciduous forests.

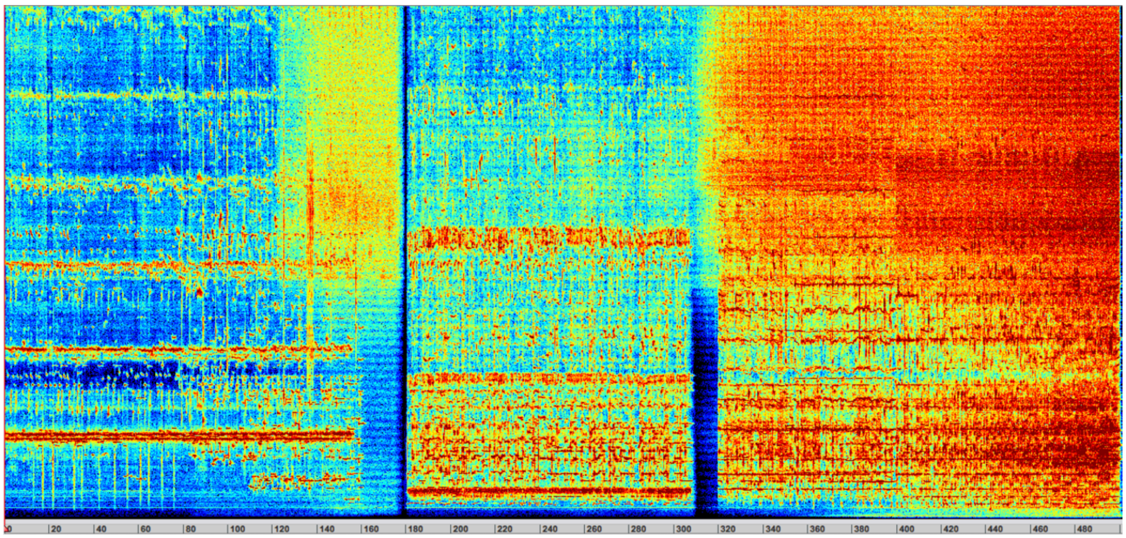


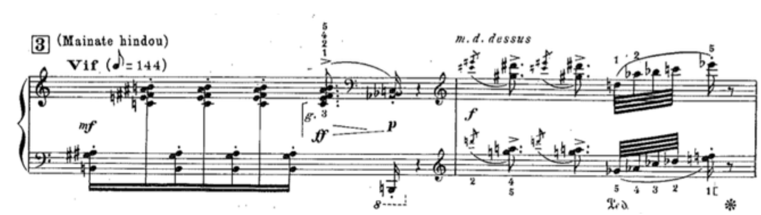
Figure 5. *Night Music*, sonogram of the recording of *Splinter Reeds* for the *Aural Shores* album

II. Environmental Sonic Translation and Analysis Techniques

The environmental sonic translation and research presented in this dissertation uses two main compositional and analytic approaches: 1) motivic and 2) spectral, with an emphasis on partial tracking. These two approaches make extensive use of computer assisted composition tools, of which the most relevant in environmental sonic translation are included at the end of this chapter.

The motivic and spectral techniques utilize specific tools and approach the sound models from different perspectives. The motivic technique focuses on pitch-rhythm, for which manual-aural transcription is a common and practical approach. The spectral technique focuses on timbral qualities (harmonic/inharmonic spectra). It is important to clarify, however, that the areas of emphasis are not exclusive—the motivic technique may incorporate timbral aspects in its treatment of the models, while the spectral technique may incorporate motivic qualities in its processes. In other words, the boundaries between both approaches are not rigid: certain spectral analysis tools may be present in the motivic-syntactical approach and, vice-versa, motivic-syntactical tools may be used in a spectral sound model-based composition.

(a)



(b)

Figure 6. Two main compositional perspectives in sound-model-based instrumental composition: a. Gerard Grisey, *Partiels* (1975) –trombone's E1 sound model treated with the spectral technique. b. Olivier Messiaen, *Oiseaux exotiques* (1955) –mina bird sound model treated with the motivic technique

2.1. Motivic Technique: Manual-Aural Transcription

This technique involves transcription of motives or phrases contained in sound models and the study of their syntax. The motivic orientation emphasizes pitch and rhythm, parameters privileged in a broad spectrum of the history of music. On the other hand, timbre, which is an essential preoccupation in the spectral technique, is not systematically analyzed in the models even when it may be an important element of compositional development.

The manual-aural (Lindborg, 2007, p. 269) method of transcription consists of a composer or analyst manually notating the sound model based on their hearing. This technique is often assisted by computer technologies such as sonograms, pitch detection, and partial tracking, as will be shown below.

Two representative composers using the motivic technique/manual-aural transcription to translate sound models into musical language are Olivier Messiaen and François-Bernard Mâche. Their work based on sound models is mostly centered on transcriptions of animal sounds, frequently derived from listening to the models through tape recordings and manually notating their musical features. Olivier Messiaen's catalogue includes a significant number of pieces derived from birdsong sound models that were notated—mostly traditionally and determinately—either in situ or listening to tape recordings (see 1.4.).

F.B. Mâche expanded his field of sound models to several other animals besides birds, human language, and the geophony. The timbral complexity and noisy features of several of the sound models featured in his music led him, since an early stage of his work, to use spectral analyses to assist his manual-aural transcription process. "We know that as early as 1964 he consulted a spectrogram, so it may be a reasonable assumption that much of his work was done through careful listening and manual transcription, aided by spectrographic frequency information." (O'Callaghan, 2015, p. 236) His notation oftentimes includes chords with very detailed microtones, which is a natural result of consulting spectrographic information, as well as indeterminate procedures, especially when the sound models present a high degree of complexity and noise.

In her comprehensive article on Mâche's complete work, *Mâche, the demiurge of sounds and the poeta doctus*, Márta Grabócz provides a list of the main types of models employed by Mâche:

1. Transcription of poems by encoding the sounding or linguistic (phonetic, syntactic) structure into music (*Safous méle*, *La peau de silence*, *Canzone III, IV*)
2. Sequence and/or juxtaposition of natural sound elements, human tongues, insects, mammals, amphibians, birds, in various groupings in different works (*Rituel d'oubli*, *Rambaramb*, *Korwar, Temes nevinbür*, *Kassandra*, *Uncas*)
3. Various bird songs: (*Korwar*, *Naluan*, *Sopiana*, etc.)
4. The sounds of the sea and the four elements of nature: (*Marad*, *Amorgos*, *Kassandra*, *Quatre phonographies de l'eau*)
5. Sampled or electronically simulated sounds of birds, insects, amphibians or human beings (*Aulodie*, *Hyperion*, *Uncas*, *Aliunde*, *Temboctou*). (Grábocz, 1993, p. 142)

Mâche uses different kinds of notation depending on the characteristics of the models, from traditional notation (models with defined pitch contours and high degree of harmonicity) to text instructions for improvisation (models with highly complex pitch content and/or high degree of inharmonicity).

Figure 7.
a) Mâche, Traditional notation

b) Mâche, Pitch-indeterminate/rhythm-determinate notation

Other composers who have done important sound-model-based composition and environmental sonic translation derived from the manual-aural approach include René Lussier (*Le trésor de la langue*, 1989) and Magnus Lindborg (*TreeTorika*, 2006). These pieces use spoken language as their sound model, analyzing and developing its motivic and syntactical components through a hybrid of the manual-aural approach and the studio or computer-based tools. René Lussier's method for *Le trésor de la langue* combines the manual-aural approach and the performers' live responses to mimic the played-back sound model. (F. Frith, personal communication, November

14, 2012) In this sense, it is a combination of a sound-score and a notated score. Magnus Lindborg, on the other hand, combines the manual-aural transcription method with advanced computer assisted composition tools in order to generate *TreeTorika*. His decision is informed by, in his words, “imprecisions” found in AudioSculpt’s spectral differencing methods and, on the other hand, more accuracy found when “using the ears.”

> The Max/MSP patcher (…) supports the transcription process. A syllable is visually and aurally determined by selecting a portion in the waveform~ window. While it loops, the user decides the pitch either by relying on the fiddle~ estimation or by checking the sound against a note on the keyboard (if necessary adjusting it by a quartertone).” (Lindborg, 2007, p. 268)

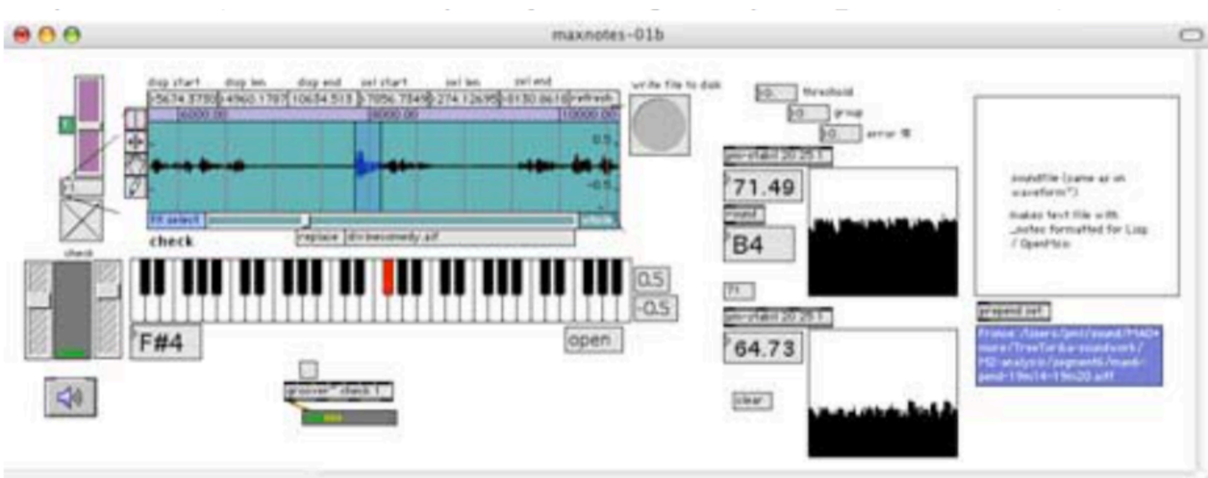


Figure 8. “Aural transcription assisted by a simple Max/MSP tool. At this point, fiddle~ estimates the pitch as B4, but the ear says it is F#4. The amplitude for each note was determined automatically, by fiddle’s velocity detection.” (Lindborg, 2007, p. 270)

Lindborg’s observation regarding the comparison between the ears and the computer assisted analysis leads to an important consideration regarding the psychoacoustics and objective/subjective dimensions in the perception of sound models, not only from the human listening perspective, but also from the perspective of the tools that we use for analysis. The closest approximation to the features of the sound model may be reached through a series of combinations of manual-aural transcription processes and computer assisted features.

My pieces, *Night Music* (4.2.2.) and *Under the Sea Ice* (4.2.3.), are aurally-manually transcribed with the aid of AudioSculpt and SPEAR. These tools provided detailed information regarding the microtonal pitches and timbre-harmonies of the stridulating insects for *Night Music* and bearded seal sound models for *Under the Sea Ice*. Decisions regarding the spectral centroid of discrete points of the melodic contours as well as the rhythmic and notational choices—rhythmically proportional with detailed microtonal information—were informed aurally.

Figure 9. Manual-aural transcription of *Under the Sea Ice* proportional notation and determined microtonal pitches

2.2. Spectral Technique

This technique, based on a static snapshot of the acoustic information of any given sound, is as broad as the definitions for the word used to describe it—"spectrum". In the context of the spectral school composers, described above, the partials of a given sound are manipulated in various ways. Filtering, amplitude modification and frequency transposition are some of the most common processes applied to spectrally analyzed sound.

In *L'Esprit des Dunes* (1994), Tristan Murail composes with sound models taken from traditional music of Mongolia and Tibet. From the latter, samples from monastic rituals including monks' chanting, dung chen trumpets, and a Jew's harp are used as sound models. From Mongolia, the model is the khöömiy, which is a technique that enables a single singer to produce a melody and a drone simultaneously by highlighting successive harmonics of a low sung fundamental. These sounds are analyzed with a tool created for additive synthesis through dynamic oscillators, developed by Guillermo Garcia at IRCAM. The tool allowed Murail a finer representation of the models. Evaluation and modification of these analyses was performed with the PatchWork program. In addition, Murail used an interface for controlling the analysis parameters and a second order filter type synthesizer in Max-FTS (predecessor of MSP) that allows control of 40 filters in real time. The main tools used in *L'Esprit des Dunes* are: (1) filtering one or more components of a spectrum by reducing or increasing its amplitude; (2) fractionating a spectral region; (3) smoothing the frequencies or dynamic components; (4) temporal modifications of the spectra (stretching and compressing); (5) distortion of the frequencies; and (6) stereo spatialization of each partial. (Daubresse et al., 2000, p. 77)

As an example of the manipulations performed by Murail, Joshua Fineberg demonstrates the natural and artificial distortions of the dung chen sound model. While the original spectrum of the dung chen already presents "a significant amount of distortion" (Fineberg, 2000, p. 122),

> [A] 'compression' of the partials can easily be observed here by looking at the shrunken octaves between partial 1 and 2, as well as 3 and 4. In the second section of that piece, the composer amplifies that natural distortion with the addition of varying amounts of artificially added distortion (Figure 11). These newly generated Spectra form the opening progression of the section. (2000)

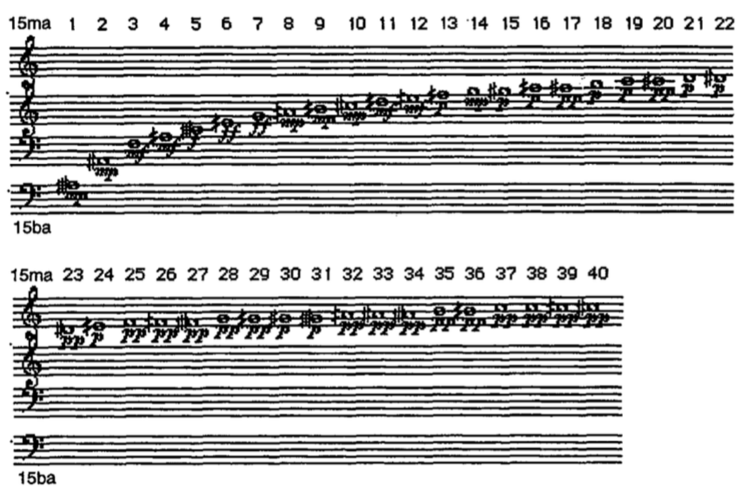


Figure 10. Spectrum of the Tibetan dung chen trumpet (Fineberg 2000, p. 122)

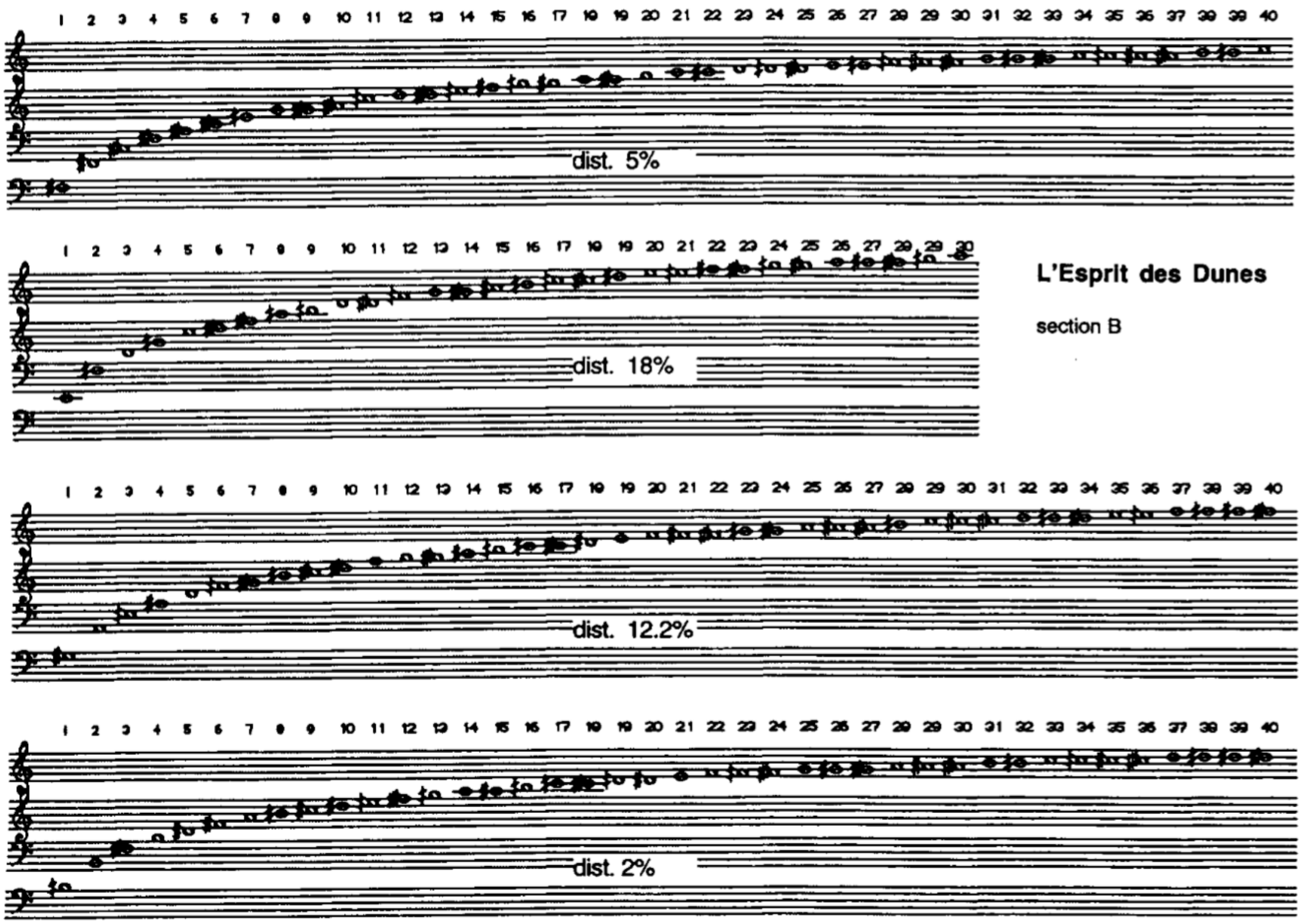


Figure 11. Artificial distortion of the partials of the dung chen sound model (Fineberg 2000, 123)

Important examples of the spectral technique are found in the work of Kaija Saariaho, who explores the "sound/noise axis to create musical tension and to replace the dynamic function of harmony." (Saariaho, 1987, p. 93) This concept, analogous to the concept of Murail's harmonic/inharmonic spectra and his techniques of compositional development discussed above, is applied both in the micro- and macro-level of composition for the development of both musical phrases and of form.

In *Lichtbogen* (1985-86, for ensemble and live electronics), pure and noisy sounds of the cello are analyzed by a computer. The analyses of the transitions within the sound/noise axis derived in the harmonic material of the piece. The main sound models consist of two contrasting techniques: (1) "the rich and noisy sounds of the cello obtained by increasing the pressure of the bow to produce a multiphonic sound" (Saariaho, 1987, p. 129); and (2) a string harmonic (natural or artificial). Saariaho derives two sound models from two types of transition: either increasing the force of the bow whilst approaching the fingerboard or sliding from one harmonic to another. "These transitions are then analyzed by taking samples from different phases in the sound." (1987, p. 129)

In addition to her notion of sound/noise as a formal procedure, Saariaho devised a multi-dimensional network in which each parameter –harmony, texture, dynamism, shifts of electronically-related color– develops under its own "curves of tension." In this way, Saariaho compensates for "the absence of large-scale tensions within the harmonic material." (1987, p. 129)

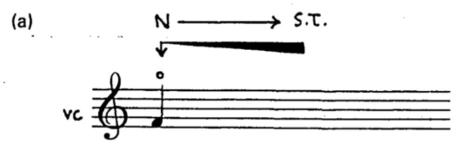


Figure 12. a) sound model: gradual increase in pressure (Saariaho, 1987, p. 129)

b) sound model: string harmonic glissandi

2.2.1. Partial Tracking

This technique stems from the spectral technique; however, it has not been exclusively used by spectral composers. The partial tracking technique involves a 'time-frequency representation', in which "the sound is sliced up, by an analysis time window, into successive instants." (Pressnitzer & McAdams, 2000, p. 37) This is achieved through the short-term Fourier transform, widely known as the Fast Fourier Transform (FFT). This procedure contrasts with the Fourier transform in that the latter limits the representation of the spectrum to frequency, providing no temporal information of the sound.

In this computer-assisted transcription technique, the composer's task is to limit or select the type of the "time-frequency" information generated. This task may rely on the ear or on a specific concept. Through the FFT analysis, the computer tracks the frequencies that constitute a sound as it evolves in time, which is a similar task to that of the composer or analyst who listens to the sound and then notates it in paper. The main difference lies in the large amounts of data afforded by computers. Composer James O'Callaghan mentions that "Arguably, the efficiency of the software-assisted approach affords a greater degree of detail, a greater 'accuracy' in the transcriptions, and engenders an aesthetic not possible in the pre-digital era." (O'Callaghan, 2015, p. 236)

Composer Fabio Cifariello-Ciardi uses this transcription technique in his work *Ab*, for nine instruments, based on a "segmentation and the morphological analysis of various recordings of speaking voices. The aim of this work is to throw light on typical attributes, such as inflection, rhythm and expressive emphasis (…)." (Cifariello-Ciardi, 2008, p. 131) This is done through the aid of "partial tracking analysis of a sample of the voice of Ghulam Nabi's father and the automatic transcription obtained from a software program written by the author [Cifariello-Ciardi] with the aid of OpenMusic." (p. 131)

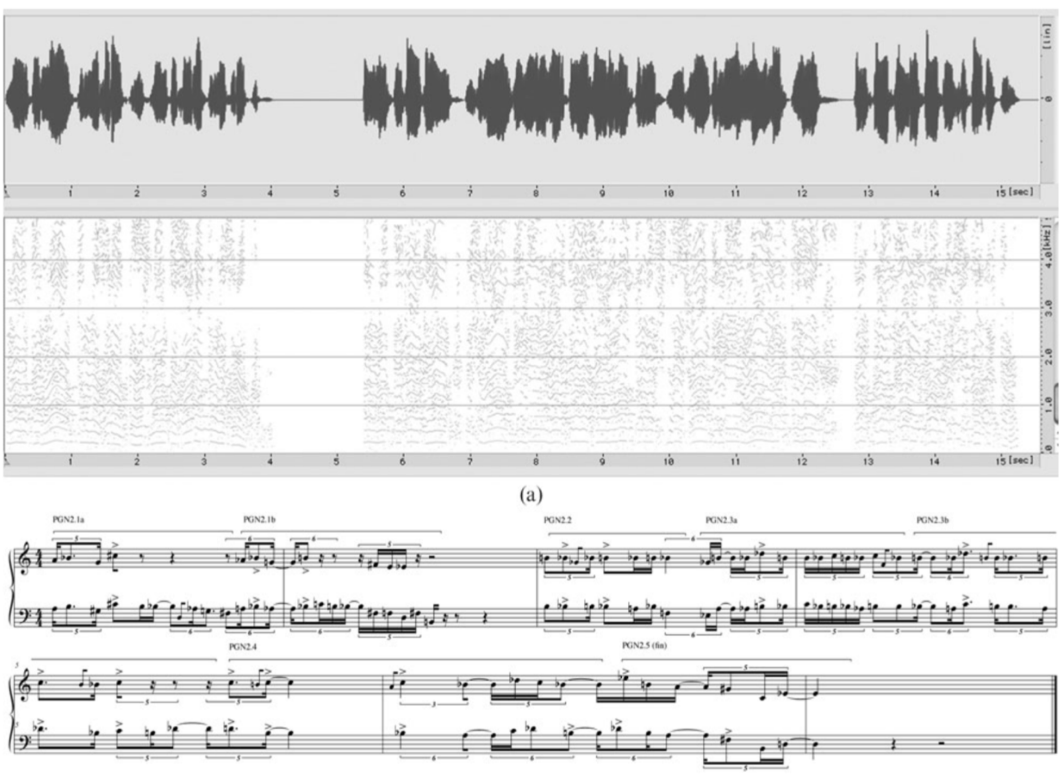


Figure 13. Cifariello Ciardi's partial tracking transcription for *Ab* (Cifariello-Ciardi 2008, 132)

Peter Ablinger's *Quadraturen IV ("self-portrait with Berlin")*, is a six-movement piece derived from a spectral analysis of six field recordings of Berlin soundscapes. Ablinger focuses on a translation of the sound model that features, in his words, a "*broken continuity*, the digital reconstruction of sound and time." This notion is applied to translate the sound models in a schema that abolishes the continuum and replaces it with the grid-like structure of the FFT. Arguably, the sound model is constituted not only by the Berlin soundscape, but by the time-frequency analysis that represents it. In this sense, *Quadraturen IV* is a commentary on the limitations of human—and therefore, technological—perception and abilities to reproduce the "complexity of reality."

Figure 14. Analysis of a Berlin soundscape with an analysis time grid of 377.5 milliseconds. Each stem in the pitches represents a pulse at 377.5 ms., equivalent to 158 bpm. (Ablinger, personal communication)

My piece, *Waves Break Aural Shores* (2018), for saxophone quartet and electronics, applies AudioSculpt and SPEAR softwares' partial tracking to a variety of angles—or *aural shores*—of the sound of the ocean, mainly: (1) highpass filtered/lowpass filtered; (2) high/low frequency resolution; (3) high/low partial decibel threshold, which may be understood as a more detailed filtering technique. The partial tracking results in broadly two kinds of notation derived from OpenMusic: (1) highly saturated and rhythmically complex (Figure 15a); and (2) long rhythmic values of continuous and slow microtonal oscillation (Figure 15b). The latter notation was not fully realized by AudioSculpt and OpenMusic. The process was finalized by manually drawing the glissando-like continuous microtonal oscillation described by the partial tracking prior to xml notation conversion in OpenMusic.

Figure 15a. *Waves Break Aural Shores,*
highly saturated and rhythmically complex notation (each bar is 4" long)

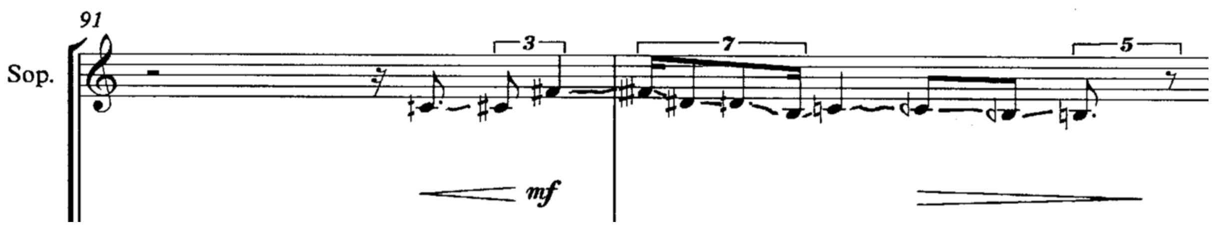


Figure 15b. *Waves Break Aural Shores*
long rhythmic values of continuous and slow microtonal oscillation (each bar is 4" long)

2.3. Computer Assisted Composition Software

The nature of the spectral approach is characterized by a "rationalism of material, already espoused by Messiaen, which can be described and therefore constructed (…) by objective procedures—free from subjective content." (Daubresse & Assayag, 2000, p. 68) Twenty years later, this claim may be modified—nuanced—as current considerations in the sciences and humanities confirm the unavoidable participation of the subject in any "objective" endeavor in varying degrees. Instead of emphasizing freedom from subjective content, it is perhaps more accurate to point towards the spectral approach's orientation towards quantifiable data as compositional foundation. Due to this orientation towards quantifiable data, "spectral composers appeared as a catalyst which would considerably aid the emergence of CAC (computer assisted composition)." (2000, p. 68) The following are the most prominent CAC resources (in chronological order) in the field of sound-model-based instrumental composition, mostly developed by IRCAM (Institut de Recherche et Coordination Acoustique/Musique) researchers since the late 1980's.

AudioSculpt (Picasso, 1995-2013) generates time/frequency analysis displayed through spectrum or sonogram and may be used for analysis and sound processing.

Max MSP (Zicarelli, 1997) may have several applications in the context of sound-model-based instrumental composition, from transcription of text files derived from audio signals into xml, to pitch detection for manual-aural transcription.

OpenMusic (Agon, Assayag & Bresson, 1998-2020) is a "superset" of PatchWork (Laurson, Duthen & Rueda, 1993), which is one of the first programs using visual programming. It responds to the question of "how can a dynamic sonic process (sound models), in which multiple parameters evolve in time, be transcribed in discrete units and symbols compatible with instrumental notation?" (Daubresse & Assayag, 2000, p. 68).

SPEAR–Sinusoidal Partial Editing Analysis and Resynthesis (Klingbeil, 2006-2018) is an application for audio editing, synthesis and analysis (representing a sound with many individual sinusoidal waves with time varying frequency and amplitude).

Orchidée (Carpentier & Tardieu, 2008) was designed for computer-assisted orchestration. It provides a set of algorithms and features to reconstruct any time-evolving sound model with a combination of acoustic instruments, given a set of psychoacoustic criteria.

III. Analysis of Representative Work in the Field

3.1. Korwar (François-Bernard Mâche): An Approach to Sound Model-Based Instrumental Music

François-Bernard Mâche is one of the most representative composers who have analyzed nature's sonic characteristics and used the emerging musical features to write music for live instruments and tape. Using geophonic, biophonic and anthropophonic sound models both as an indispensable tool for material extraction and as the tape component of his pieces, Mâche's music typically features a strikingly closely knit connection between the instrumental part and the pre-recorded part—the sound model. In one of his most important writings, *Music, Myth and Nature,* he claims that "One of the most universal practices of musical creation is the use of sound models. The imitation of animal sounds by primitive ethnic groups of hunters is likely to give us the most direct image of the common source of myth and of music." (Mâche, 1983, p. 36) The detail in Mâche's rhythm and pitch transcriptions of his various sound models, as well as the imagination with which he translates sonic features such as density and timbre, are faithful reflections to his statement: "What seems to me interesting is precisely to try to bring together poetics and theory, and to show the advantages that there are in advancing an aesthetic project on the basis of a harmony with natural data". (1983, p. 166)

This section consists of an analysis of a microcosm of Mâche's work: *Korwar*, for harpsichord and tape (Mâche, 1972). This piece manifests Mâche's interdisciplinarity from various angles: musicology, zoo-musicology, linguistics, acoustics, and composition, all fields to which the composer committed throughout his life. *Korwar* features sound models derived from recordings of animals, the Xhosa language—a branch of the Niger-Congo languages, official in South Africa and Zimbabwe—, rain, and pre-recorded sounds of the harpsichord itself. These various sounds were analyzed by the composer acoustically, musically, and linguistically, and further transcribed and translated into a piece of ~14'45" in duration.

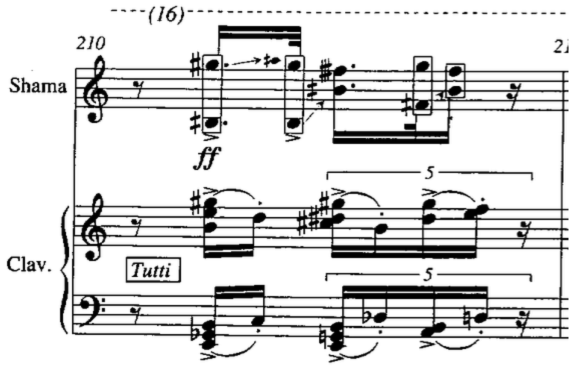


Figure 16. *Korwar*, bar 210

3.1.1. Analytic categories

Korwar features multiple events with varying degrees of harmonicity and inharmonicity (pitch–noise relationships), determinacy and indeterminacy, and synchronicity between sound models and performer. The objective of this analysis is to establish a system that describes the flux between such categories. Acknowledging the relevant analysis of Kippelen, which uncovers *Korwar's* melodic and harmonic relationships based on the timbral qualities of the sound models (Kippelen, 2005, p. 41), this analysis overlooks the motivic and harmonic elements of the piece in order to focus on the relationship between timbre, performance strategies (such as determinacy and indeterminacy), and sound model. The analysis provides some keys to the principles under which F.B. Mâche translates the sound models into music notation.

The piece is examined through three analytic categories, each consisting of two poles and in-between states that are in constant flux throughout the piece. These categories point towards striking compositional angles from which Mâche combines the material and generates contrast:

Timbre			
pitch	noised pitch	pitched noise	noise
Determinacy–Indeterminacy			
determinacy	indeterminate pitch	indeterminate rhythm	indeterminacy
Synchrony–A-synchrony			
synchrony	semi-synchrony	a-synchrony	

In sum, this analysis focuses on: (1) the musical materials derived from each sound model in terms of the three analytic categories; (2) the combinations among these categories and their in-between states, as well as their formal implications.

The analytic categories draw meaningful relationships between the sound models and choices of orchestration and notation, as will be explained in 3.1.4.

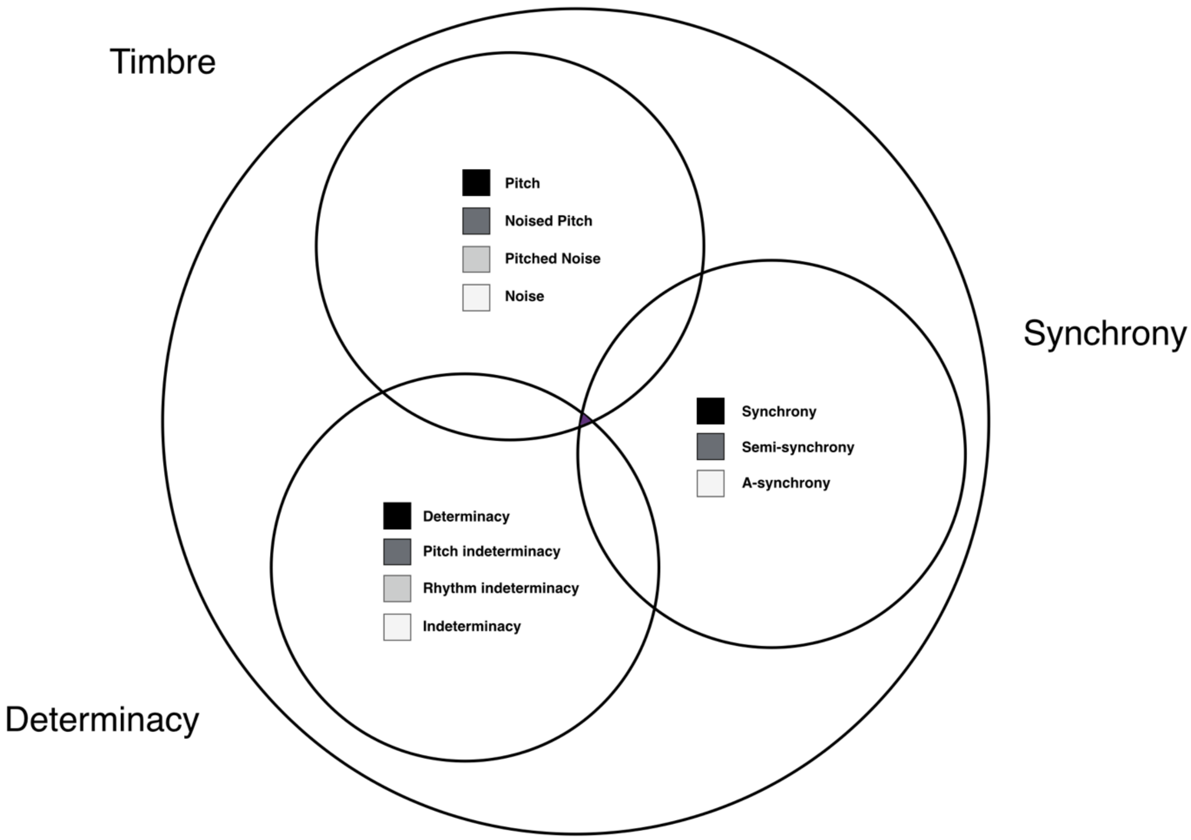


Figure 17. Analytic Categories diagram

3.1.2. Sound Models in Korwar

The tape part for *Korwar*, titled *Agiba* by the composer, is not only used by Mâche in this piece, but also in two other works: *Rambaramb*, for instrumental ensemble and tape (1972) and *Temes Nevimbür,* for two pianos, percussion and tape (1973). The three pieces are considered the "Melanesian cycle", given the fact that the titles are all in languages from Oceania. With minor variants, the sequence of the sound models is identical in the tape part for the three pieces. The sound models used in the *Agiba* tape part are the following, in order of appearance in the piece:

- Xhosa: a Nguni Bantu language, a branch of the Niger-Congo languages. It is the official language In South Africa and Zimbabwe. Xhosa is a tonal language characterized by click consonants.
- Shama: bird inhabiting densely vegetated habitats in Southeast Asia
- Frogs
- Starling: bird most common in Europe, also resident in South Africa
- Boar
- Guanaco: camelid native to South America
- Old Boar
- Killer Whale
- Rain
- Pre-recorded harpsichord (exclusively in the tape part for *Korwar*)

3.1.3. Timbral Profile of the Models

The following is the profile[2] of each of the sound models, based on the pitch/noise axis described in 3.1.1. The models characterized by pitch are situated on the top, while those characterized by noise are on the bottom. The in-between states are shaded accordingly.

Shama 2	**Pitch**
Guanaco	**Pitch**
Starling	**Noised Pitch**
Xhosa	**Noised Pitch**
Frogs	**Noised Pitch**
Killer Whale	**Noised Pitch**
Boar	**Pitched Noise**
Shama 1	**Pitched Noise**
Old Boar	**Pitched Noise**
Guanaco	**Pitched Noise**
Shrimp	**Noise**
Rain	**Noise**

[2] This profile was generated based on an aural approach. A potential feature in the analysis could be the use of virtual fundamentals in order to determine the models' varying degrees of harmonicity/inharmonicity.

3.1.4. Sound model transcription

The degree of detail with which F.B. Mâche incorporates the sound models into the *Korwar* score is evident in his use of a staff dedicated exclusively to the transcriptions of the tape. The entirety of the contents of the tape part are featured in notation above the harpsichord part. This is a common practice throughout his work in different periods. Works like *Amorgos* (1979) and *Canopeé* (2003), include staves with detailed transcriptions of the tape/sampler parts. This serves two purposes: 1) logistically, it aids the performer in the extremely difficult task of synchronizing with the material; 2) aesthetically and conceptually, it re-signifies the natural sound models as music by representing them with musical language.

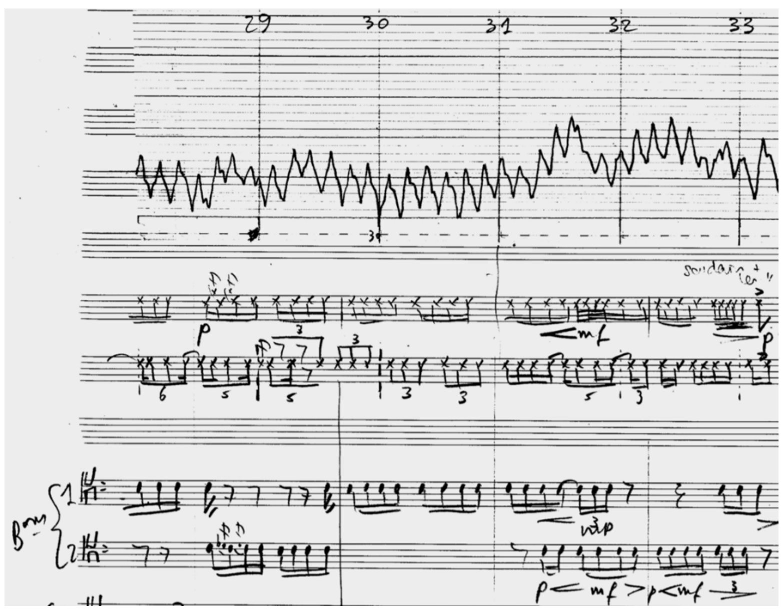


Figure 18. *Amorgos*, page 5, manuscript (fragment), courtesy of F.B. Mâche. Top to bottom: 1. Amplitude display of the sound model; 2. Rhythmic transcription of the model; 3. Bassoons 1 and 2.

In the domain of pitch in *Korwar*, it is of particular interest the manner in which the timbral profile—the pitch to noise ratio—of a sound model determines the notation used in its transcription. When the sound model presents a high degree of harmonicity and the pitches are clearly audible, the pitch notation is traditional, specific, and includes microtones. The complexity of the timbres, especially of the birds, reveals Mâche's use of spectrograms to determine highly specific pitch information. On the contrary, when the sound model presents inharmonicity and noise is predominant, the pitch notation is left ambiguous, without clefs, occasionally indicating ascending or descending general motion. As an example, the figures below present the most significant notations used for the models' transcriptions in *Korwar*. While the "starling" in fig. 19b, characterized by "pitch", is transcribed with regular note-heads and detailed microtonal pitches, the "boar", characterized by "pitched noise", is notated with "x" note-heads and without a clef, in order to observe an undefined pitch notation. Further, "rain" and "shrimp" in fig. 19d, characterized by "noise", are notated with dotted lines as a means to indicate a lesser degree of pitch than the round or crossed note-heads. In *Korwar*, the dotted lines indicate the highest degree of noise, present in the rain, shrimp and old boar sound models.

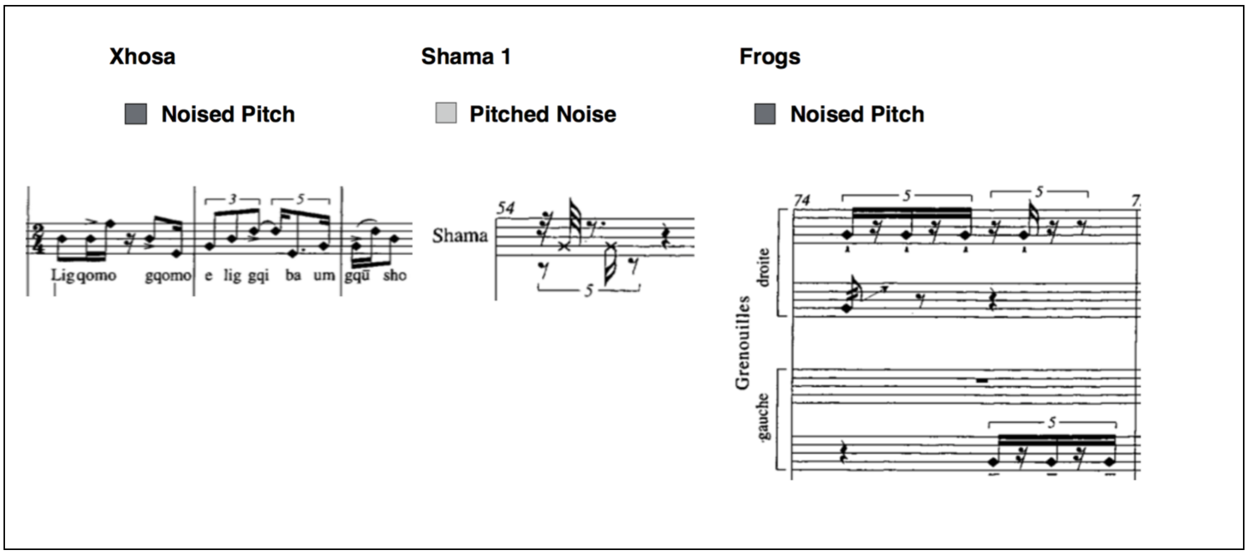


Figure 19a. *Korwar*, notations used for the models' transcriptions

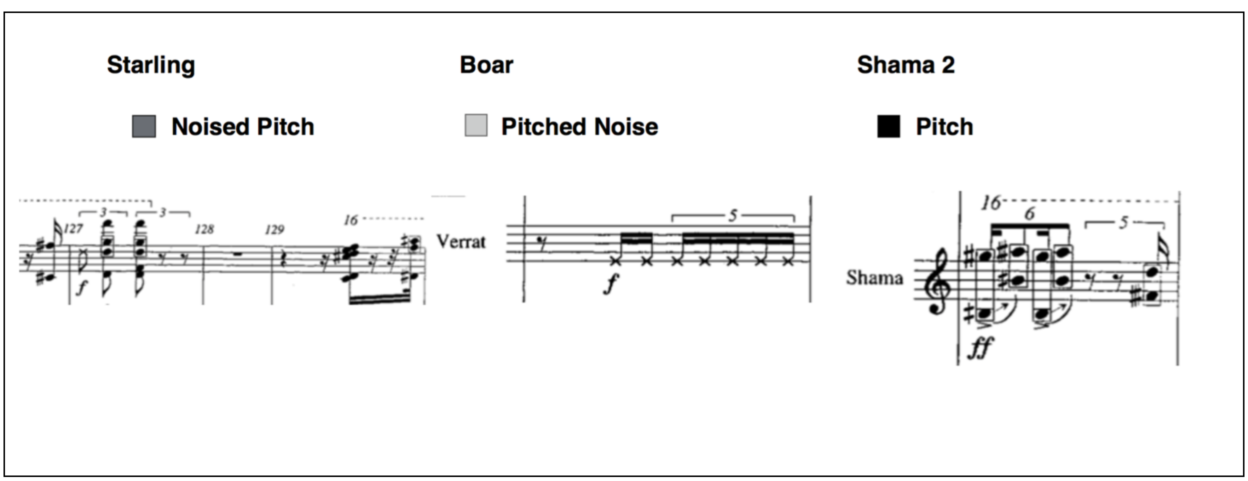


Figure 19b. *Korwar*, notations used for the models' transcriptions

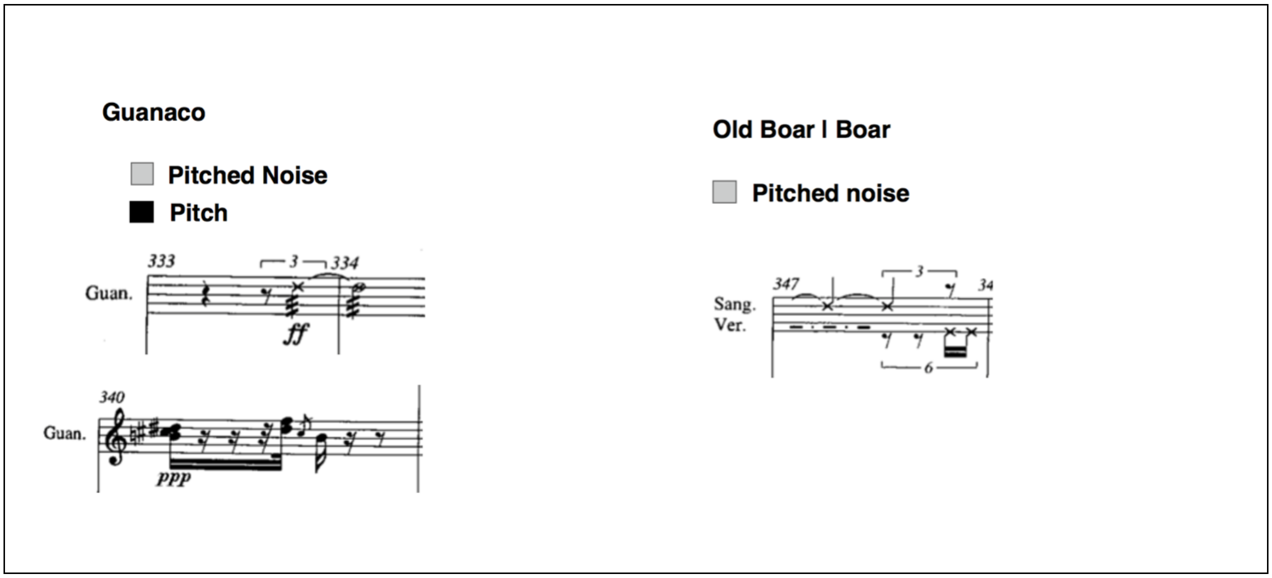


Figure 19c. *Korwar*, notations used for the models' transcriptions

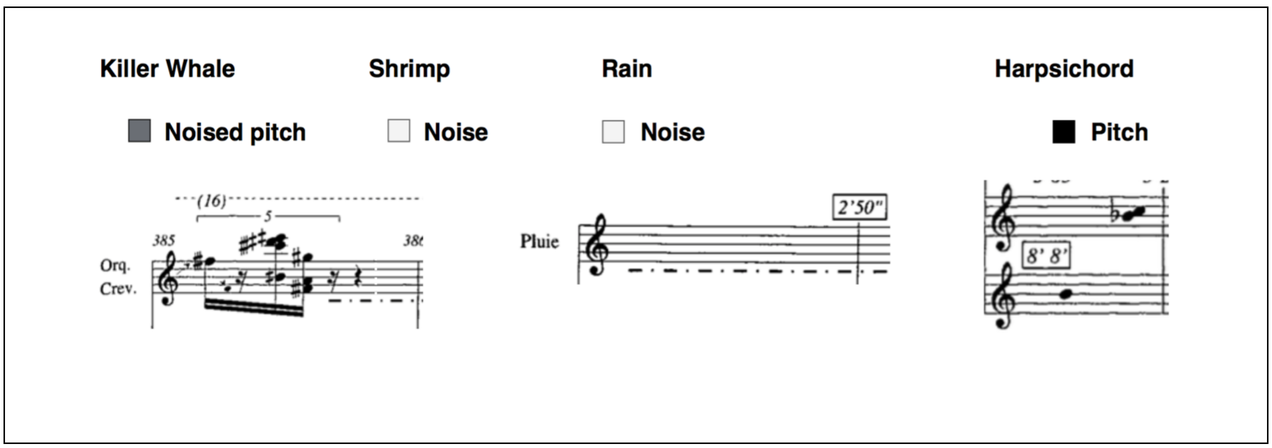


Figure 19d. *Korwar*, notations used for the models' transcriptions

In the rhythm domain, the score generally presents fully detailed traditional notation due to the rhythmic intelligibility of most of the sound models, with the exception of the "boar", "shrimp" and "rain". The "boar" emits bursts of noise in rapid iterations, varying the rhythmic tuplet groupings slightly (i.e., from sextuplets to quintuplets at ~112 beats per minute). The noisy quality of its sound often blurs the attack and therefore the rhythm, resulting in a cacophony nearly impossible to notate rhythmically. On the other hand, the tumultuous multiplicity of the "shrimp" and "rain" sound models derives in similar undefined attacks. These are the only instances in which Mâche uses an indeterminate notation to describe the rhythm.

3.1.5. Sound model-based notations for the harpsichord

While the timbral and rhythmic profiles of the sound models determine the notations used in the transcriptions (3.1.4), these profiles and transcription notations determine the notation and performance strategies for the harpsichord. As mentioned earlier, there are meaningful correspondences between the categories of timbre, determinacy and synchronicity, and these inform the notation used for the sound model transcription and the performance strategies assigned to the harpsichord. This section presents the main harpsichord notations applying the analytic categories combinations. It is worth noting how, while there is a general consistency between the sound model and the timbre/determinacy/synchrony profile, Mâche generates a variety of combinations within the correspondences between sound model and harpsichord. This is especially true when the sound model's timbre has some degree of noise. In the following figures, the notation variants are presented following the overarching order of 1. Pitched; 2. Noised pitch; 3. Pitched noise and 4. Noise.

3.1.5.1. Pitched sound models

Timbre (model): Pitched
Timbre (hpd[3]): Pitched
Determinacy: Determinate
Synchrony: Synchronized

Figure 20a. *Korwar*, pitched sound model notations, variant 1

[3] hpd: abbreviation for "harpsichord"

Timbre (model): Pitched
Timbre (hpd): Pitched
Determinacy: Indeterminate rhythm
Synchronicity: A-synchronized

* Performance note: "the harpsichord plays the left hand on the downward stems, the right hand on the upward stems, and enchain the rhythms avoiding any cyclic cell and repetition of more than two notes or identical chords"

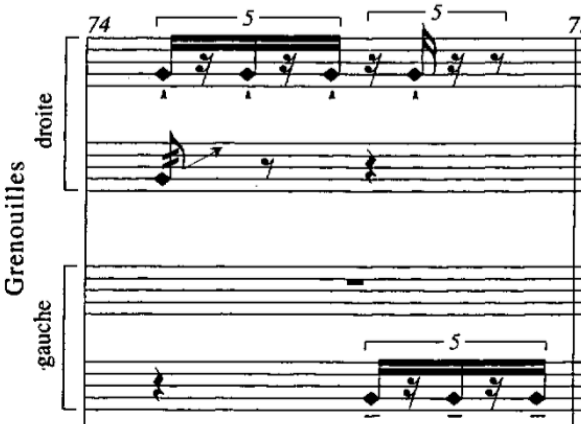
Figure 20b. *Korwar*, pitched sound model notations, variant 2

Timbre (model): Pitched
Timbre (hpd): Noised pitch
Determinacy: Determinate
Synchronicity: Synchronized

Figure 20c. *Korwar*, pitched sound model notations, variant 3

3.1.5.2. Noised pitch sound models

Timbre (model): Noised pitch
Timbre (hpd): Noised pitch
Determinacy: Indeterminate pitch
Synchronicity: Semi-synchronized



Clav. 

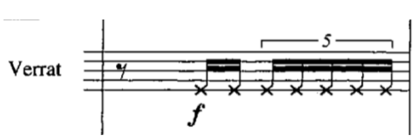
(indeterminate material for the harpsichord)

* Performance note: “improvisation on clusters of very nervous rhythms such as: (see fig. above), independent of the tape part, with silences. Clusters on all the keyboard, those of the medium will be able to be repeated quickly. Little by little the rhythms will align with those of the tape part, until ending almost synchronous at the end of the sequence.”

Figure 21. *Korwar*, noised pitch sound model notations

3.1.5.3. Pitched noise sound models

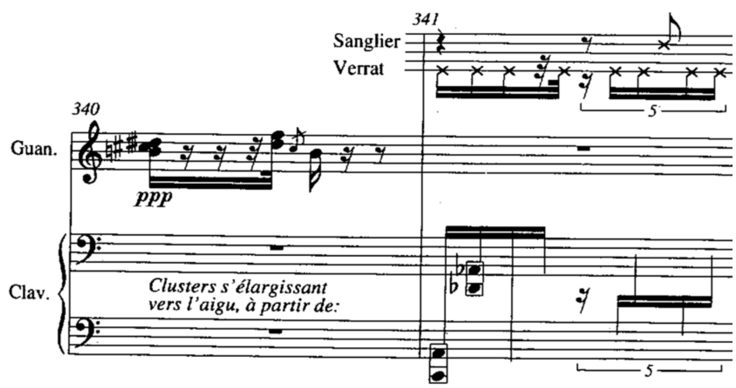
Timbre (model): Pitched noise
Timbre (hpd): Pitched noise
Determinacy: Indeterminate
Synchronicity: Semi-synchronized



* Performance note: “improvisation analogous to the beginning (bar 74, previous figure), without creating cycles, in unequal sixteenth notes at the same tempo, with some brief silences, with gradual expansion of the aggregates towards the low and the high (epilation of the clusters)”

Figure 22a. *Korwar*, pitched noise sound model notations, variant 1

Timbre (model): Pitched noise
Timbre (hpd): Pitched noise
Determinacy: Indeterminate pitch
Synchronicity: Synchronized

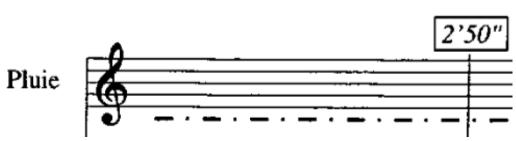


* Performance note: “clusters widening towards the higher register, starting from: (see fig. above)”

Figure 22b. *Korwar*, pitched noise sound model notations, variant 2

3.1.5.4. Noise sound models

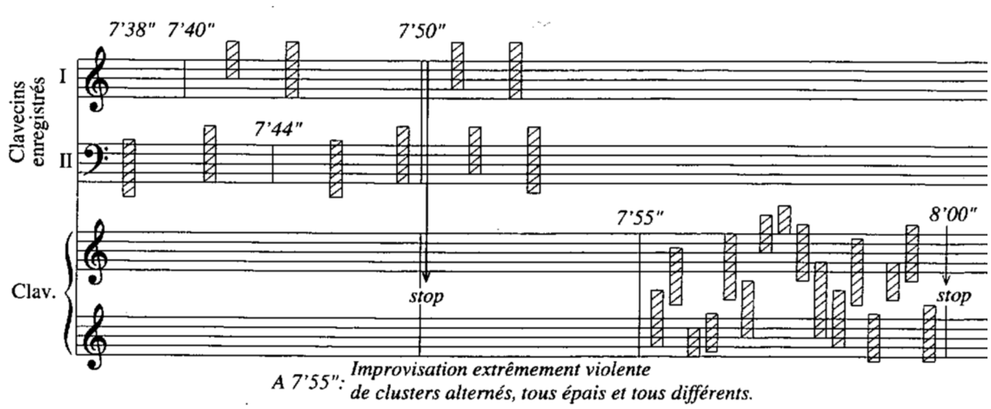
Timbre (model): Noise
Timbre (hpd): Noise
Determinacy: Indeterminate
Synchronicity: Semi-synchronized



* Performance note: Improvisation analogous to the tape part on the mute keyboard (noise of the jacks alone) and some shocks on the mating pedal, without ringing a note

Figure 23a. *Korwar*, noise sound model notations, variant 1

Timbre (model): Noise
Timbre (hpd): Noise
Determinacy: Indeterminate
Synchronicity: A-synchronized



* Performance note: "extremely violent improvisation with alternating clusters, all dense and different"

Figure 23b. *Korwar*, noise sound model notations, variant 2

3.1.6. Sound model-based registers for the harpsichord

The register choices for the sound models' transcriptions throughout the piece suggest correspondences between the model's timbral features and the assigned harpsichord registers. Predominantly, models that present a timbre with a higher pitch to noise ratio (,) are transcribed into the higher registers of the harpsichord. Inversely, models characterized by noise (,) are transcribed into lower registers. Detailed timbral analyses of the low, medium and high registers of the harpsichord (and of most harmonic instruments) are not necessary to easily perceive the degree of pitch or noise particular to each of the instruments regions. A spectral analysis of a pitch in the low register of most harmonic instruments will contain a higher amount of audible dissonant harmonics (i.e., harmonics 16–32) than a corresponding pitch in a medium or high register. Bar 341 (Figure 24), where the harpsichord presents clusters over a major sixth and a perfect fifth in low register contiguous chromatic fundamental notes C2 and Db2, the resulting timbre is mostly noise due to the aforementioned acoustic properties of low register pitches. This choice of register is a natural correspondence to the noisy timbral profile of the boar (*verrat,* in French) sound model, as described above in 3.1.3.

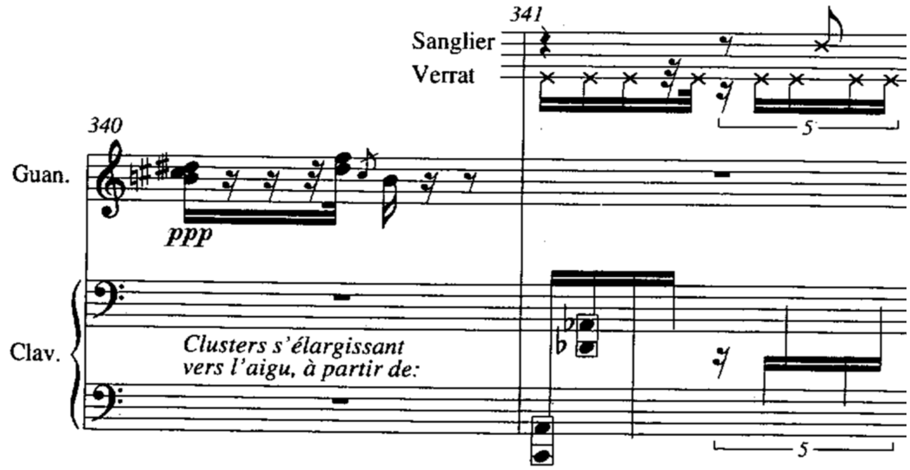
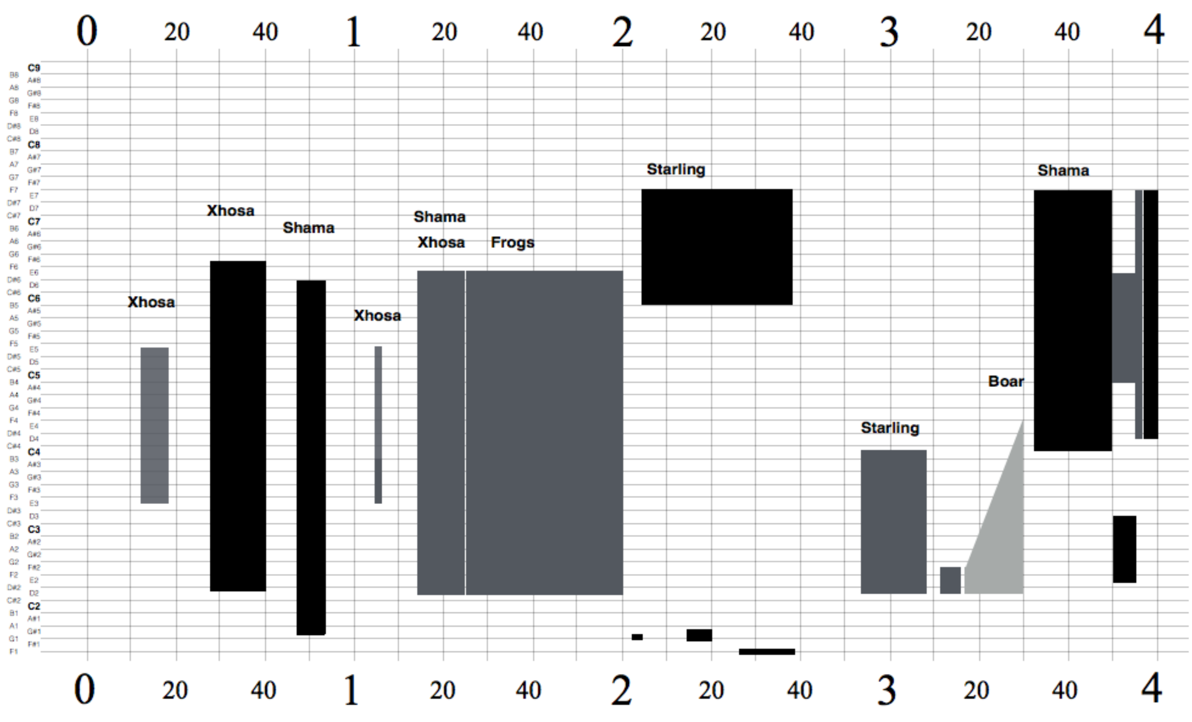
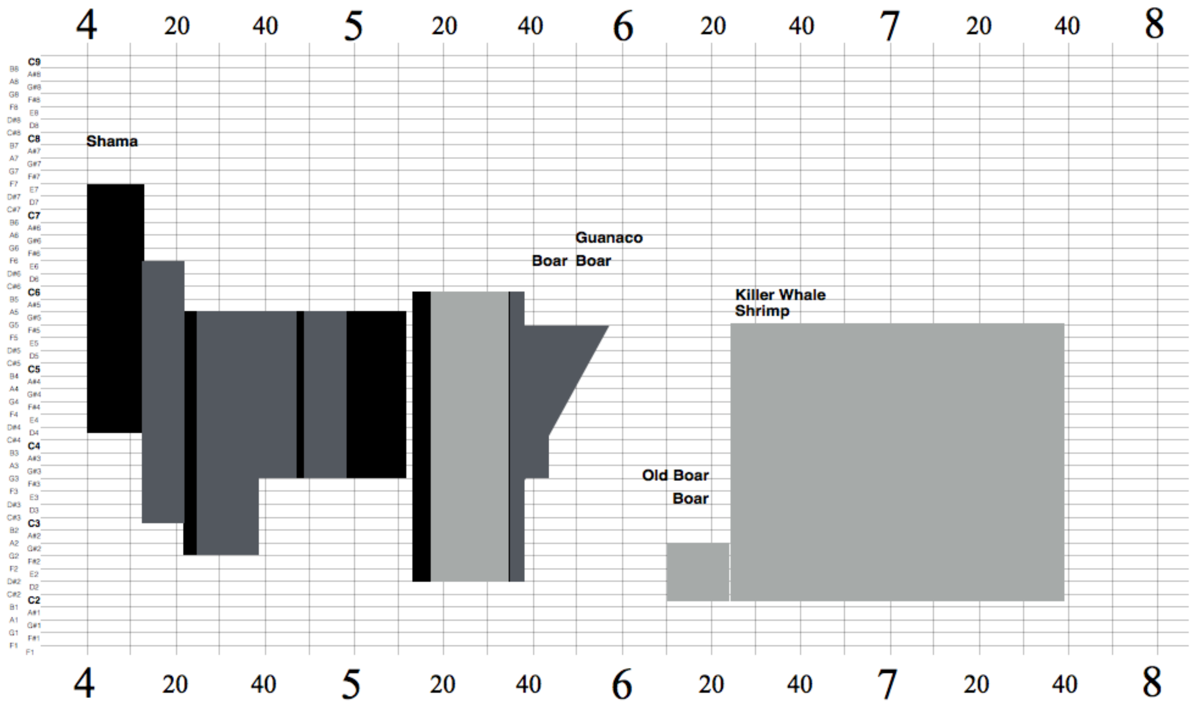
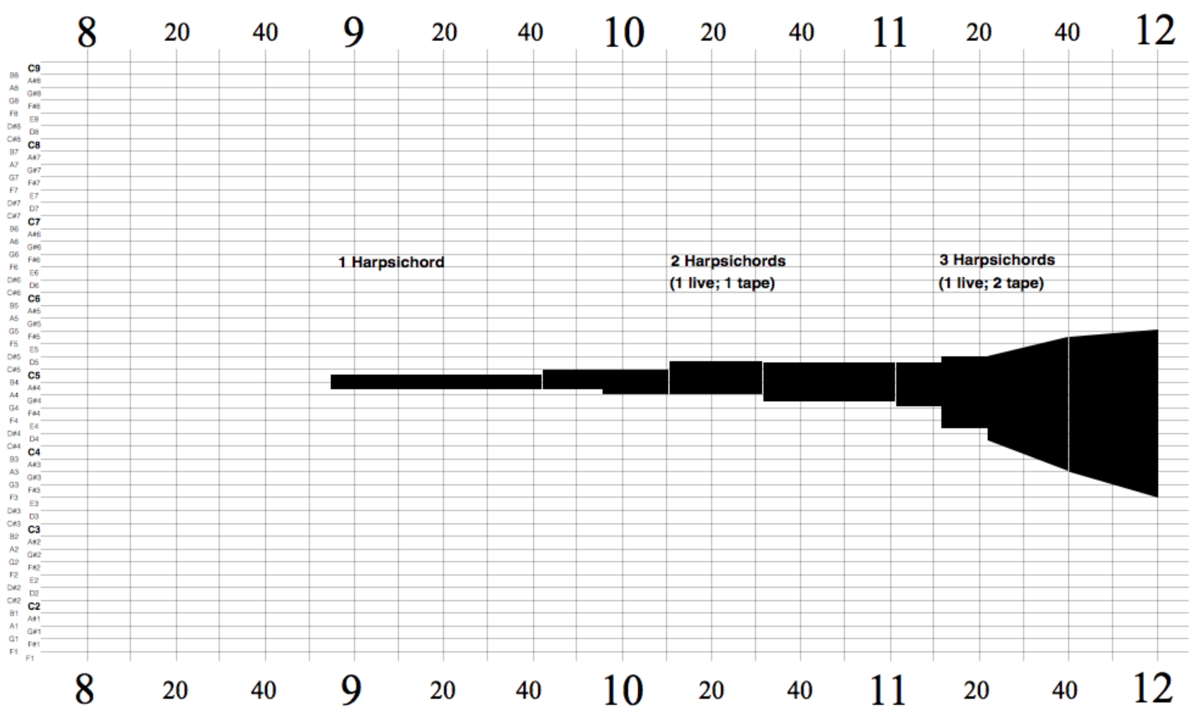


Figure 24. *Korwar*, cluster production of noise

Figure 25, below, is the register/timbre map for the harpsichord in the entirety of *Korwar*. Above the timbre color-coded register areas is the sound model for the material of the harpsichord. This map shows the systematic correspondence between the sound models' timbral profiles, as explained in 3.1.2., and the register of the harspichord. Note the general correspondence between ◻ (noise) with lower registers and ■ (pitch) with higher registers.




4 20 40 5 20 40 6 20 40 7 20 40 8
Shama
Guanaco
Boar Boar
Killer Whale
Shrimp
Old Boar
Boar


8 20 40 9 20 40 10 20 40 11 20 40 12
1 Harpsichord
2 Harpsichords
(1 live; 1 tape)
3 Harpsichords
(1 live; 2 tape)

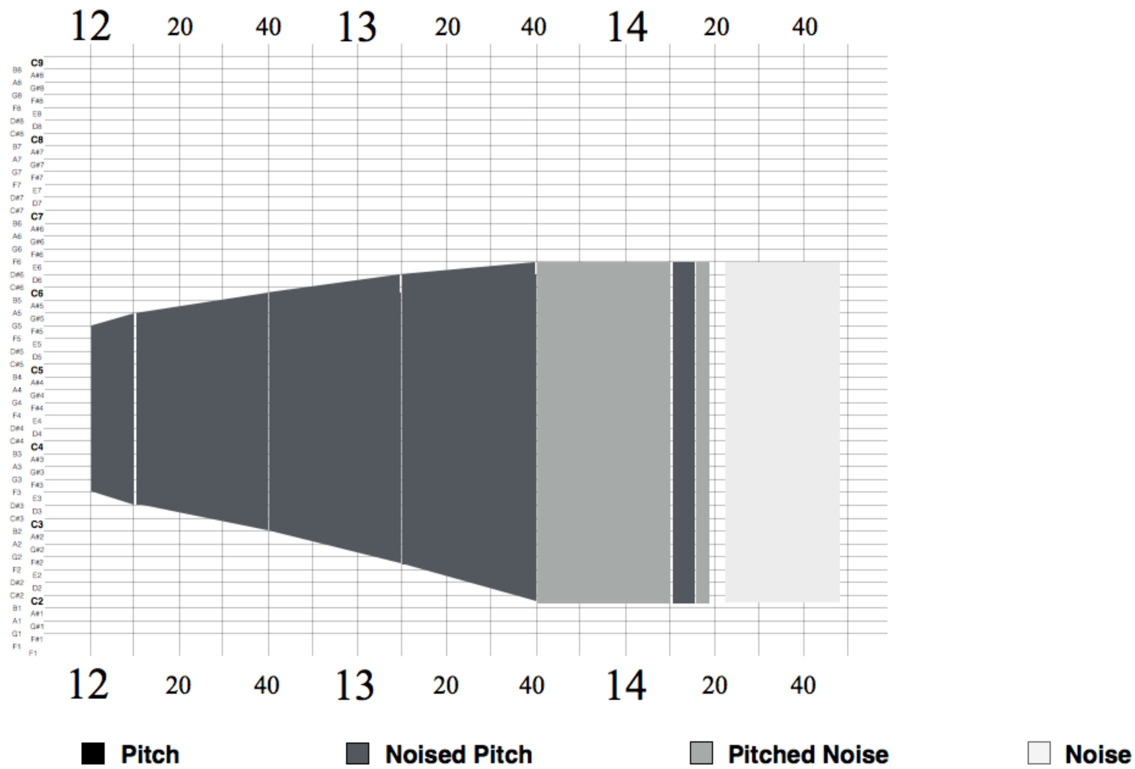


Figure 25. *Korwar*, register/timbre map for the harpsichord

3.1.7. Form

Korwar is constructed in eight sections (I–VIII), distributed into two compound sections (ABC, ABC'), which in turn consist of timbral regions A, B, and C, and A', B' and C'. Each section (I–VIII) is characterized either by a sound model or by a group of sound models that share either timbral or textural features. From the lens of this analysis, both the formal schema of the piece and the development of its materials find their driving force in the timbral domain. For this reason, sections I–VIII are distributed into compound sections consisting of timbral regions: ABC and ABC' (Figure 26). The timbral regions are expressed with A (noised pitch/pitched noise), B (pitch), C (noise), scaled to a smaller dimension in the last section of the piece, ABC'.

The example below (Figure 26) presents the overarching compound sections (ABC, ABC') and each section (I–VIII) with its corresponding shade-coding according to the timbre category (i.e. the pitch-noise axis). In ABC, the organization of the sound models and their translations into music notation for harpsichord present an overall progression from gray ■ (noised pitch) to black ■ (pitch) to white □ (noise). A striking feature of the piece is the beginning of ABC', section VIII, where the sound model ceases to be a zoological, geological or anthropological-vocal sound model. Two pre-recorded harpsichords become the sound models. The same progression from noise to pitch and back to noise is observed in this section, which is a synthesis and microcosm of ABC (sections I–VII). The second compound section (ABC'), with a material that has not been introduced in the previous sections and that contrasts from all the previous events in the piece, has a duration of more than 1/3 of the piece. This is an enigmatic statement that F.B. Mâche is inscribing within the form of the piece. An interpretation of this is that the music has found a synthesis between the sounds of nature and the sounds produced with human means. While the first section of the piece (ABC) presents a juxtaposition of synchronous and a-synchronous interactions between the zoological/geological/Xhosa language sound models and the harpsichord, the second section (ABC') is characterized by the absence of the sound models. However, the same timbral progression "■■□" is preserved. Is this perhaps a manifestation of Mâche's aim towards "a harmony with natural data"? This may be expressed in the last section of *Korwar* as purely human music that distills the timbral relationships of the natural world presented in all the sections preceding VIII. Preserving the timbral contour "■■□" in both sections ABC (I-VII) and ABC' (VIII), within the context of the considerable differences in the material used in both sections, is an important element that leads towards the notion of the embodiment of "natural data" in Section VIII.

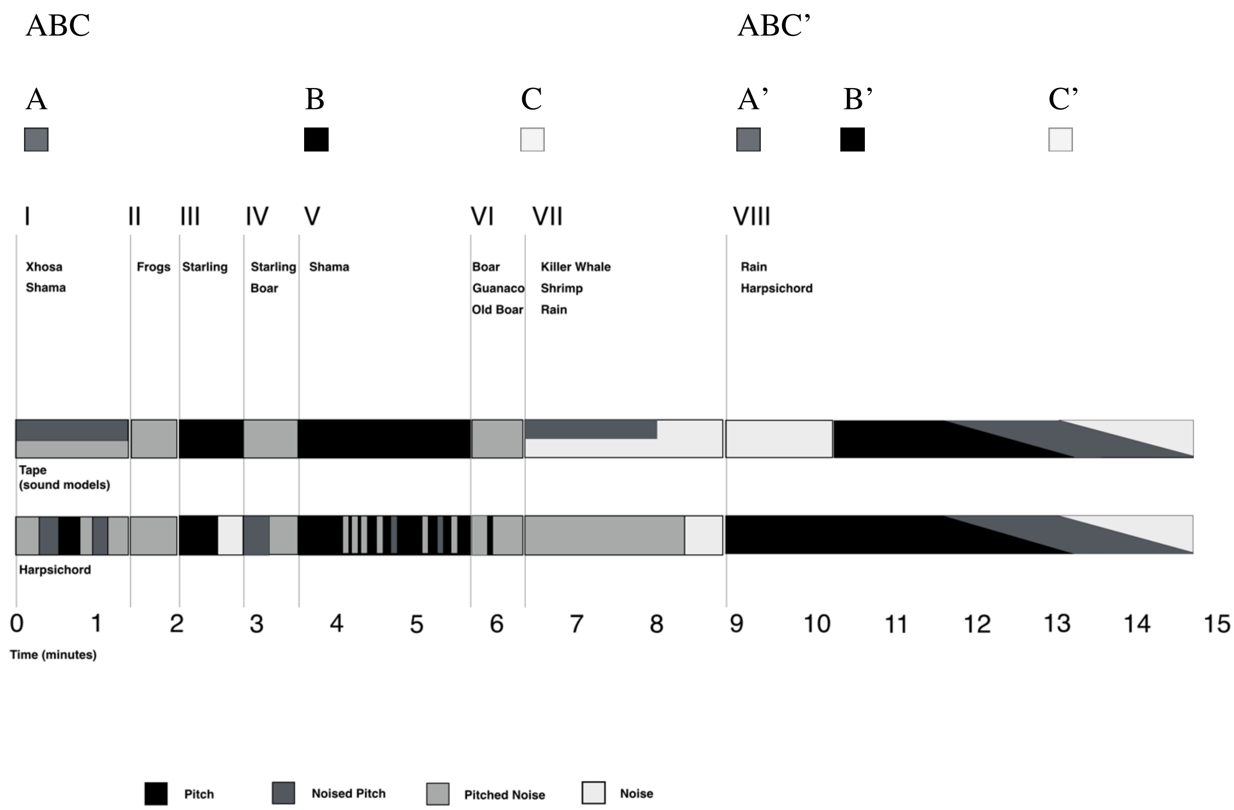


Figure 26. *Korwar*, overarching and discrete sections

The following example (Figure 27) features the more detailed correspondences between timbre, synchrony and determinacy in selected fragments of each section. The matching shades in the bottom (determinacy), middle (synchrony) and upper (timbre) levels of the columns evidence Mâche's tendency to use: 1. determinate notation and synchronicity for models with a high degrees of harmonicty; 2. indeterminate notation and a-synchrony for models with high degrees of inhamonicity. The figure also displays exceptions to this rule, such as sections VII and VIII, in which the harmonicity that characterizes the sound models of both sections are not translated into determinacy or synchrony between the model and the harspichord, as in the other sections of the piece.

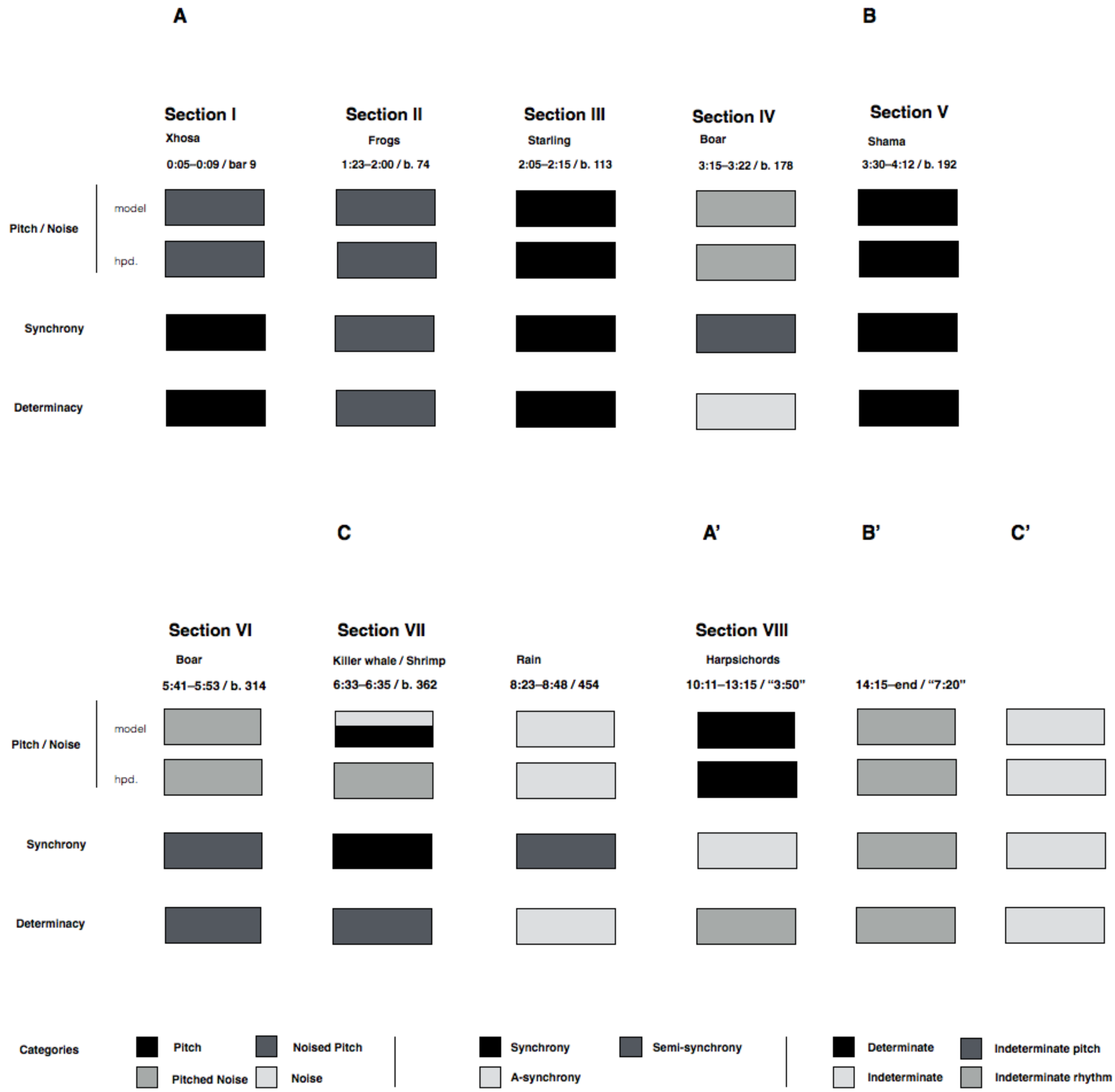
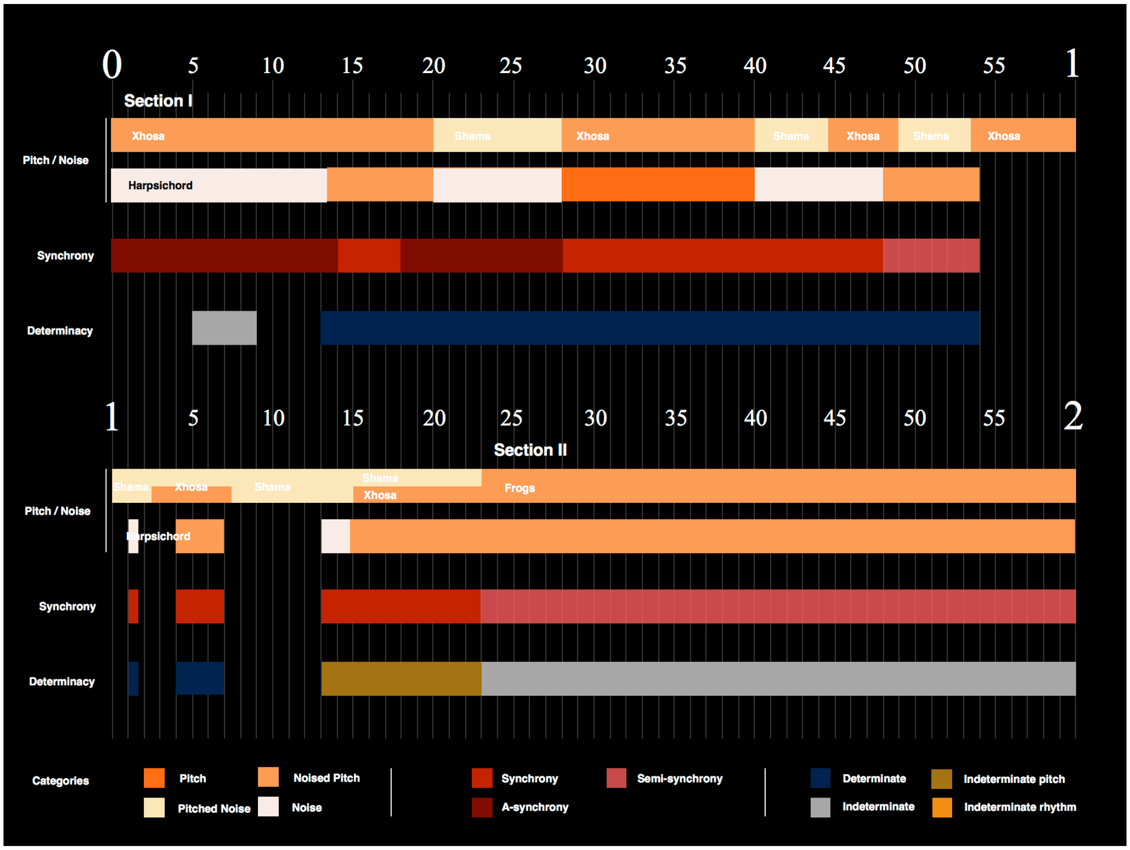
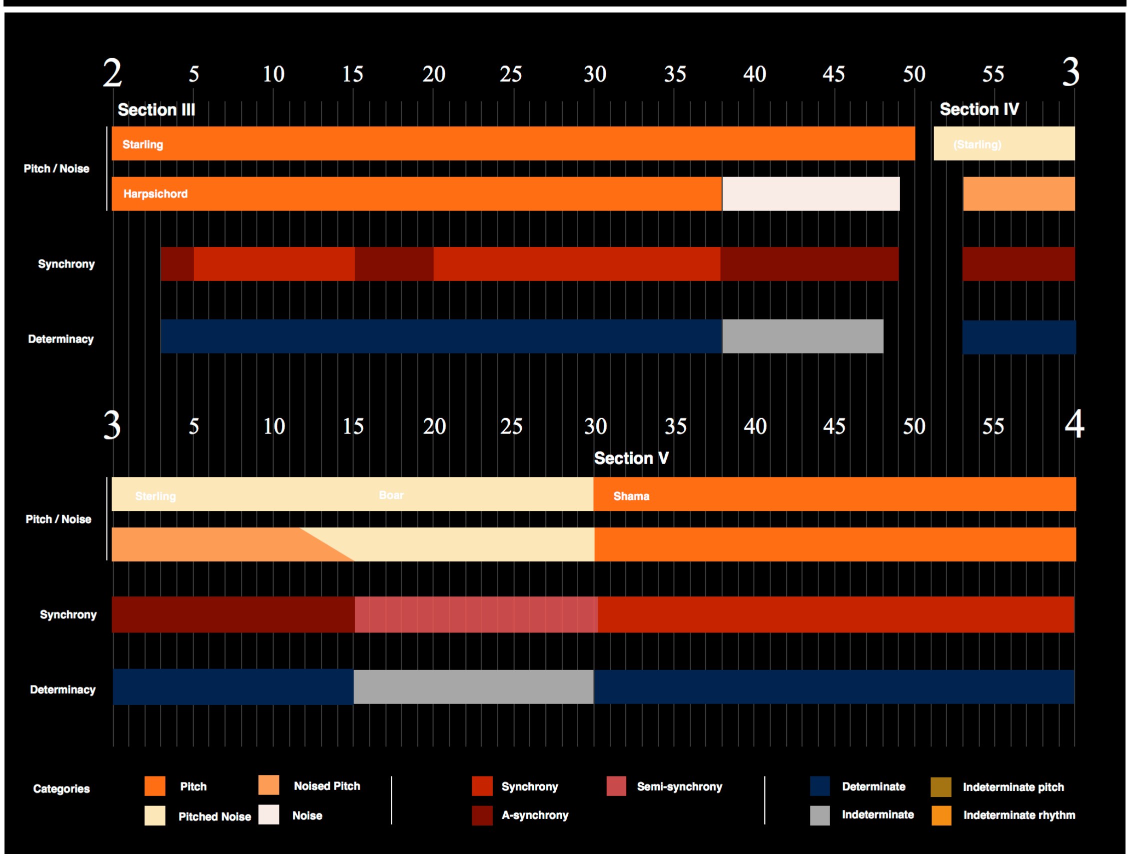


Figure 27. *Korwar*, correspondences between timbre, synchrony and determinacy within the form

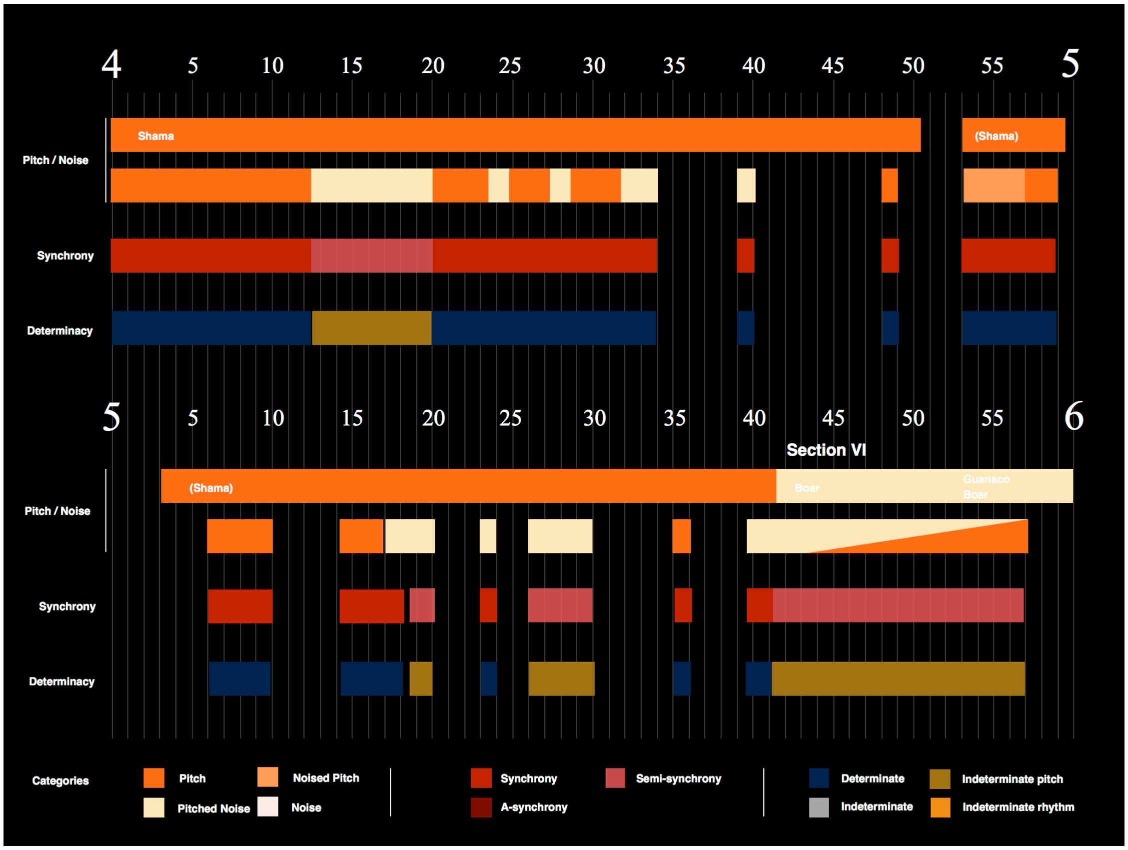
The following example (Figure 28) is a detailed description of relationships between the three analytic categories in the entire piece. While the figure above is an abstraction of such relationships for the sake of a formal notion, the figures below show the various ways in which Mâche translates the sound models, free from a methodological rigidity yet being congruent with his principles.

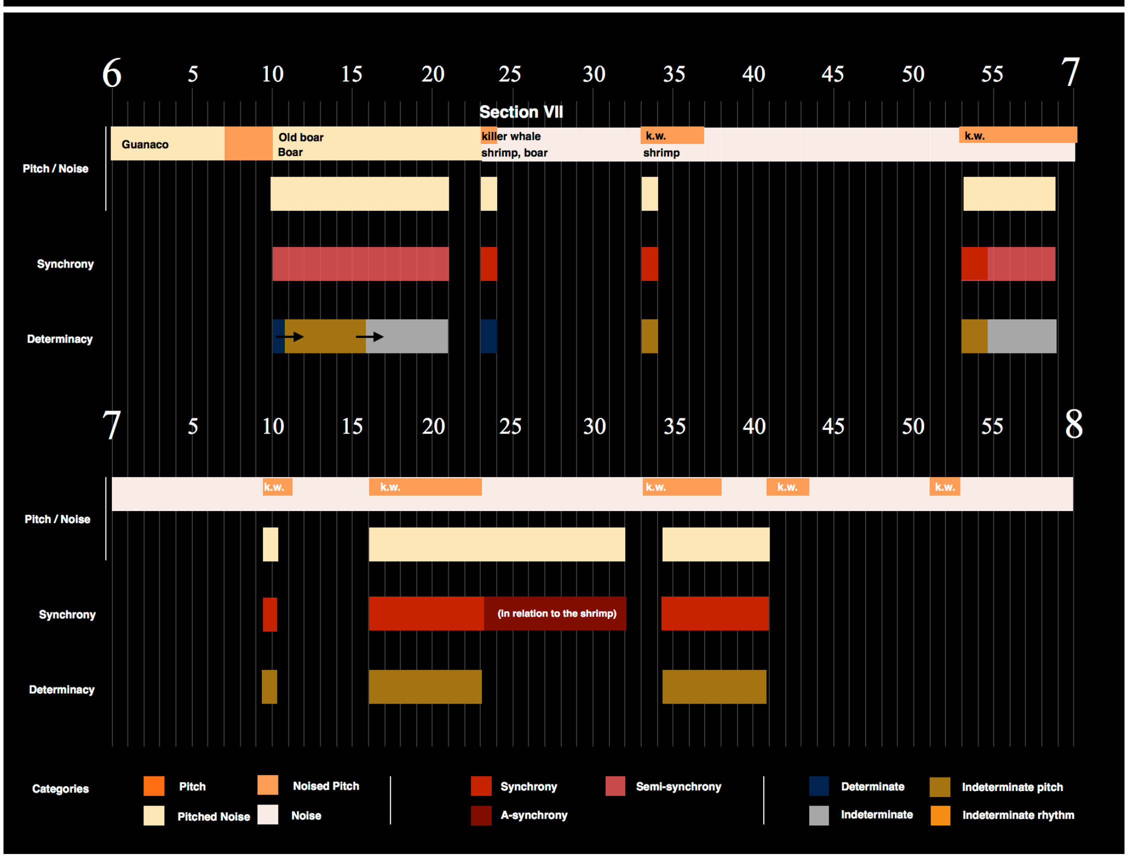


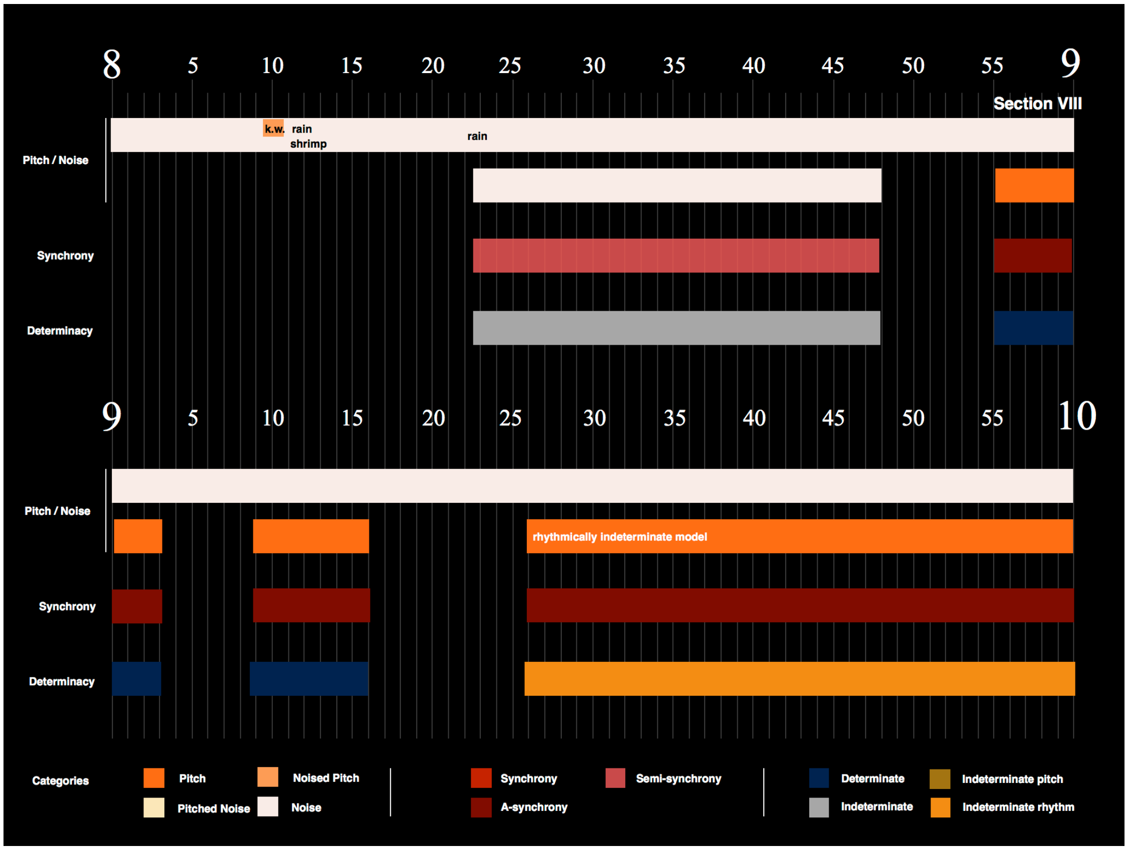
0 5 10 15 20 25 30 35 40 45 50 55 1
Section I
Pitch / Noise
Xhosa
Shama
Xhosa
Shama
Xhosa
Shama
Xhosa
Harpsichord
Synchrony
Determinacy
1 5 10 15 20 25 30 35 40 45 50 55 2
Section II
Pitch / Noise
Shama
Xhosa
Shama
Shama
Xhosa
Frogs
Harpsichord
Synchrony
Determinacy
Categories
Pitch
Noised Pitch
Pitched Noise
Noise
Synchrony
Semi-synchrony
A-synchrony
Determinate
Indeterminate pitch
Indeterminate
Indeterminate rhythm

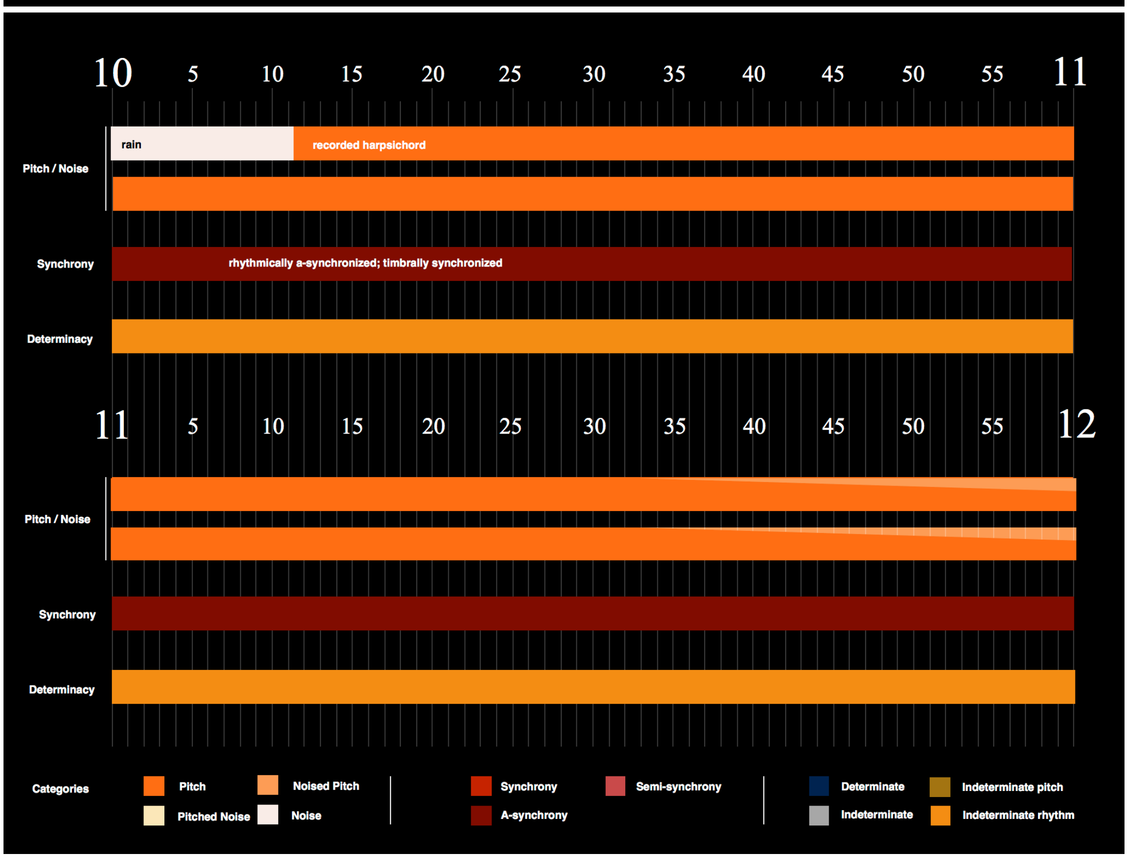


2 5 10 15 20 25 30 35 40 45 50 55 3
Section III
Section IV
Pitch / Noise
Starling
(Starling)
Harpsichord
Synchrony
Determinacy
3 5 10 15 20 25 30 35 40 45 50 55 4
Section V
Pitch / Noise
Sterling
Boar
Shama
Synchrony
Determinacy
Categories
Pitch
Noised Pitch
Pitched Noise
Noise
Synchrony
Semi-synchrony
A-synchrony
Determinate
Indeterminate pitch
Indeterminate
Indeterminate rhythm


4
5 10 15 20 25 30 35 40 45 50 55
5
Pitch / Noise
Shama
(Shama)
Synchrony
Determinacy
5
6
Section VI
(Shama)
Boar
Guanaco
Boar
Categories
Pitch
Noised Pitch
Pitched Noise
Noise
Synchrony
Semi-synchrony
A-synchrony
Determinate
Indeterminate pitch
Indeterminate
Indeterminate rhythm


6
5 10 15 20 25 30 35 40 45 50 55
7
Section VII
Pitch / Noise
Guanaco
Old boar
Boar
killer whale
shrimp, boar
k.w.
shrimp
k.w.
Synchrony
Determinacy
7
8
k.w.
k.w.
k.w.
k.w.
k.w.
(in relation to the shrimp)
Categories
Pitch
Noised Pitch
Pitched Noise
Noise
Synchrony
Semi-synchrony
A-synchrony
Determinate
Indeterminate pitch
Indeterminate
Indeterminate rhythm


8
5
10
15
20
25
30
35
40
45
50
55
9
Section VIII
Pitch / Noise
k.w.
rain
shrimp
rain
Synchrony
Determinacy
9
5
10
15
20
25
30
35
40
45
50
55
10
Pitch / Noise
rhythmically indeterminate model
Synchrony
Determinacy
Categories
Pitch
Noised Pitch
Pitched Noise
Noise
Synchrony
Semi-synchrony
A-synchrony
Determinate
Indeterminate pitch
Indeterminate
Indeterminate rhythm


10
5
10
15
20
25
30
35
40
45
50
55
11
Pitch / Noise
rain
recorded harpsichord
Synchrony
rhythmically a-synchronized; timbrally synchronized
Determinacy
11
5
10
15
20
25
30
35
40
45
50
55
12
Pitch / Noise
Synchrony
Determinacy
Categories
Pitch
Noised Pitch
Pitched Noise
Noise
Synchrony
Semi-synchrony
A-synchrony
Determinate
Indeterminate pitch
Indeterminate
Indeterminate rhythm

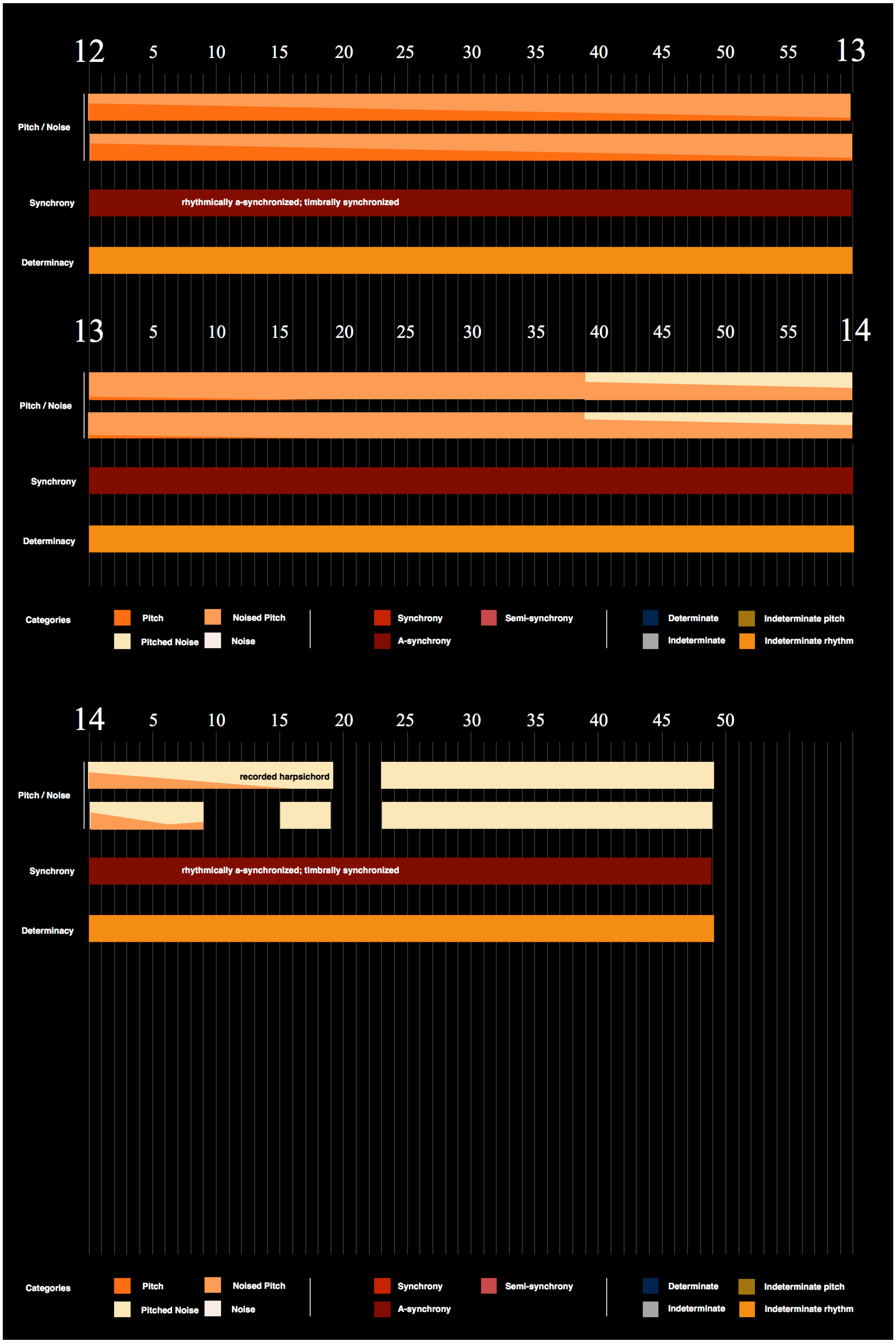


Figure 28. *Korwar*, detailed correspondences between timbre, synchrony and determinacy within the form

3.2. Quadraturen IV, “self-portrait with Berlin” (Peter Ablinger): Composing Listening

Peter Ablinger’s *Quadraturen IV (“self-portrait with Berlin”)* is a six-movement piece derived from a spectral analysis of six field recordings of Berlin soundscapes. The analysis of the six recordings becomes the score for an ensemble comprised of fl., 2 cl., 2 vl., vla., 2 vc., 3 keyboards, and CD, which performs the music synchronized to the original recordings. In his program notes for the piece, Ablinger mentions the following:

> The music becomes an observer of reality. Compared with ‘reality’, ‘music’ is defined in terms of a scanner (with horizontal rhythm and vertical pitch). To be precise, in terms of a very rough scanner which hobbles far behind the complexity of reality. But at the same time, such hobbling reflects the truth of the observation process as well as being an aesthetic phenomenon in itself. (Ablinger, Quadraturen IV, n.d.)

This analysis begins with an overview of the salient musical and conceptual features of *Quadraturen IV*. It then explores different connections of the piece as a translation of “reality” into music, with theories of perception developed by biologist Humberto Maturana, cyberneticist Gregory Bateson, and art critic Jonathan Crary. Lastly, an alternate experimental idea of a spectral analysis and translation to music for performers is provided as a reading of Bateson’s *multiple versions of the world,* Maturana’s ideas about perception in *Autopoiesis and cognition,* and Ablinger’s techniques used in *Quadraturen IV*. The proposed idea is an example of an approach to a composition methodology that incorporates simultaneous modes of perception and instrumentations of the same environmental recording. This proposed compositional methodology was the initial idea for a piece that I wrote a year after writing this analysis: *Waves Break Aural Shores*, which is explained in detail later in Chapter 4.

3.2.1. Musical features of *Quadraturen IV*

Written between 1995 and 1998, Quadraturen IV (“self-portrait with Berlin”) is part of the Quadraturen cycle of installation, electroacoustic, and concert pieces composed between 1997 and 2004 by Peter Ablinger. With regards to Quadraturen IV, Ablinger explains:

> The original material consists of 6 different microphone recordings with ambient sounds of the city. These form both the (raw) accompaniment to and basic material for the instruments, which, parallel to the recordings, play the analysis resulting from the temporal and spectral scan of the latter. Here, one might also speak of the maximum integration of instrumental sounds in environmental recordings. (Ablinger, Quadraturen IV, n.d.)

3.2.1.1. Concept: the spectral grid

The title of the entire cycle, *Quadraturen*, is the German for "squarings". In this sense, Ablinger is alluding to his process of translation of the environmental, speech, and musical recordings, in which the time and frequency of the acoustic photograph, the *phonograph*, is broken into a grid of squares that become the score. The score is then reproduced on traditional instruments, computer controlled piano, or white noise. The spectral analysis software, implemented by Thomas Musil at the Institut für Elektronische Musik (IEM) at Graz, used temporal screening—a series of rapid static analyses—which could be understood as a spectral grid. Ablinger decided to sequence the spectral screens one after the other in relation to the timeline of the original recordings. Time and frequency are dissolved into a grid of small "squares" of, for example, 1 second (time) to 1 second (interval). The result of this is, in his words, is a "broken continuity: the digital reconstruction of sound and time." This is characteristic of the *Quadraturen* pieces, and especially of *Quadraturen IV*, as the instruments do not characterize the sound of reality as a continuum but as a grid. Throughout the cycle there are varying degrees of resolution depending on the square sizes in the grid. Depending on the resolution, the degree of brokenness of continuity varies.

3.2.1.2. Time grid resolution and tempi

The resolution of the spectral and temporal grid in *Quadraturen IV* is low in comparison with that of other pieces in the *Quadraturen* cycle that are reproduced with electronic means and not performed by human instrumentalists. While the computer-controlled piano is able to reproduce up to 16 units per second with precise frequency and amplitude in *Quadraturen III –Deus Cantando*, average instrumentalists necessitate the grid to be enlarged in order for it to be playable in a score. Ablinger mentions that when “using a smaller grain, e.g. 16 units per second, the original source approaches the border of recognition within the reproduction.” (Ablinger, Quadraturen IV, n.d.) However, in an enlarged grid, which is an inevitable condition of notated instrumental music, recognition is considerably hindered and the listener is only able to perceive glimpses of similarities between the original and its musical version.

In *Quadraturen IV,* the *quadraturen–squarings–broken continuities,* are translated into the score mostly as a heterophony in which all the parts follow successions of repetitive rhythm, which are felt like pulse *ostinati*. The constant, repetitive attacks at each pulse are breaking the continuum of the environmental recordings in the same way in which digital spectral analysis fragments the audio signal into a given number of discrete units. In this sense, each attack may be understood as the division of each spectral screen that is sequenced one after the other. In other words, each attack is a square in the grid of time.

Figure 29. *Quadraturen IV,* movement I.

Throughout the six movements of *Quadraturen IV*, the resolution time grid develops and changes over time (Figure 30). Whenever there is a pulse fluctuation, mostly through rhythmic subdivisions, there is a change in the time grid and therefore in the resolution. The longer the value, the lower the resolution and vice-versa. In contrast with the way in which Ablinger treats the frequency domain’s grid in Quadraturen—which consists of semitones remaining constant as the frequency resolution throughout the piece—, the time domain fluctuations are an extremely dynamic resource for development in the piece. They imply a poly-faceted perspective of reality, with different degrees of proximity and distance to the acoustic object—the Berlin soundscape—, revealing the observer's perceptual apparatus constantly shifting.

Figure 30. *Quadraturen IV*, movement I. Simultaneous different time grids (bar 57 only).

The previous example (Figure 30), bar 57, illustrates two simultaneous time grids. The first three staves –fl., cl. 1 and cl. 2– are on a time grid at a meter of 4/4, with a tempo of eighth note = 160 bpm. This time grid is the opening grid of the piece. The remaining staves, vl. 1 and 2, vla., and vc. 1 and 2, present the new time grid—superimposed to the previous one—, playing slightly longer durations (approximately a dotted sixteenth longer) with respect to those of fl., cl. 1, and cl. 2. To be precise, the strings are now playing rhythmic values of quarter notes within quintuplets (they attack every four sixteenth notes within successive quintuplets) (Figure 31). That new superimposed time grid played by the strings is equivalent to a tempo of quarter note = 100, which becomes the general tempo in the next section of the first movement.

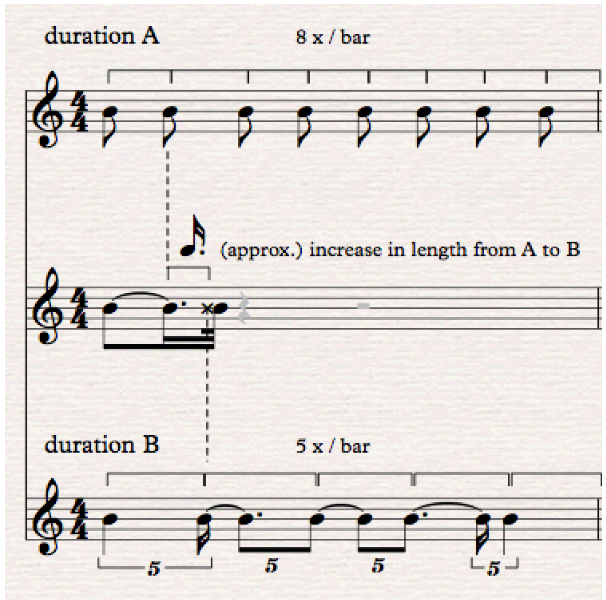


Figure 31. *Quadraturen IV,* Duration/Time grid transformation from duration A to duration B

Similar processes follow throughout the score. A new time grid is introduced by means of a new rhythmically generated pulse that is superimposed to the current one and that either works as a polytempo or gradually substitutes the previous tempo and establishes itself as the new tempo in a simpler, non-tuplet form (i.e., instead of tied tuplet values producing the sense of a pulse, a rhythmic value establishes itself as the pulse). The following sketch by Peter Ablinger (Figure 32) illustrates how this principle works in movements I through V:

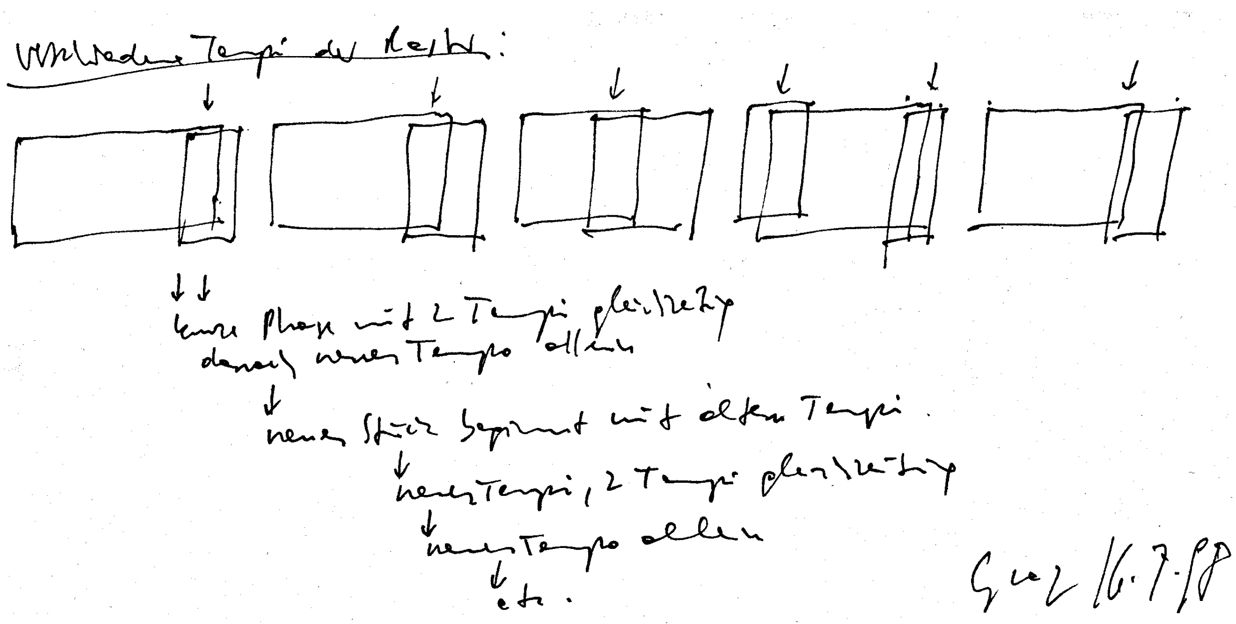


Figure 32. *Quadraturen IV*, "Time changes with its overlapping zones; drawing from note-books"
(P. Ablinger, personal communication, March 2, 2017)

3.2.1.3. Amplitude segmentation / Orchestration

The pitch domain and orchestration in *Quadraturen IV* was determined after an amplitude stratification of the partials resulting from the spectral analysis. As explained above, the partials are presented in a frequency resolution of semitones. Depending on their amplitude, the partials are situated among a spectrum of amplitude strata ranging from ***p*** to ***ff***, laid out vertically. In the first movement, there are five amplitude zones in decibels. From loudest to quietest:

ff	0.0 … –20.0 dB
f	–20.0 … –24.0 dBå
mf	–24.0 … –26.5 dB
mp	–26.5 … –29.0 dB
p	–29.0 … –31.5 dB (P. Ablinger, personal communication, March 2, 2017)

The criteria for the orchestration is mainly to assign an amplitude segment or a group of segments to specific instrumental sections of the ensemble. Ablinger explains the criteria for the first movement:

> the ***ff*** layer goes to the wind trio
> the ***f*** to the string 5-tet (quintet)
> ***mf***, ***mp*** and ***p*** goes to the 3 keyboards
> if one chord in one layer contains more voices than available instruments, I took just the loudest notes of that layer (March 2, 2017).

Below is a fragment of the spectral analysis for the opening movement of *Quadraturen IV* (Figure 33). We can observe the distribution of the partials among two of the five amplitude or dynamic regions. The document also reveals the time-grid resolution in the upper left of the image, in which we can see "8 x 377.5 ms", which shows how the 8th note is 377.5 ms long at a pulse of 160 beats per minute. Another feature revealed in the chart is the composer's choice of groupings for the 8th notes, from which the meters of the piece are derived.

Figure 33. *Quadraturen IV*, "Note Analysis" (fragment) (March 2, 2017)

The two staff groups shown in the example above (Figure 33) are the source material for the wind trio and string quintet, respectively, as explained by Ablinger. In the score, the dynamics are slightly modified for musical and orchestration reasons. The wind trio plays ***f sempre*** while the string quintet plays ***mf sempre***. The same distribution displaying the winds on the foreground and the keyboards on the background in terms of dynamics is used throughout the entire piece for the layout of the instrumentalists on stage as well (Figure 34). The loudspeakers are placed in stereo at both ends of the front of the stage.

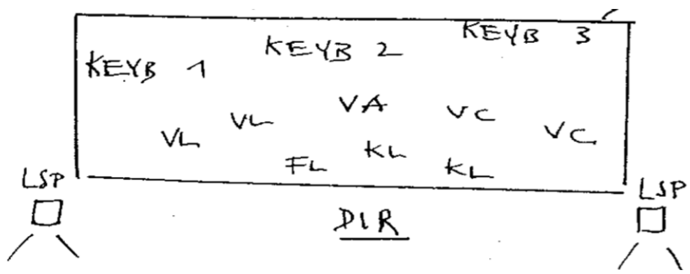


Figure 34. *Quadraturen IV*, Stage setup (Ablinger, 1995-98)

3.2.2. Translation of reality. Perceptual implications

3.2.2.1. The self-portrait

The subtitle of the piece, "self-portrait with Berlin", which presents the music as a self-portrait of the composer with Berlin, reveals a fundamental preoccupation of Ablinger: presenting the view of the observer instead of presenting the observer her/himself and that which is viewed (the object). It is "what I see (and hear) that forms the subject of the piece. In other words the portrait reveals the view of the observer (…)." (Ablinger, Quadraturen IV, n.d.) The observer is the composer who is analyzing, transcribing and translating the acoustic reality which, in the case of this cycle, is a series of field recordings of Berlin. The observer is using tools of music and electronics in order to share his view, and these tools present conditions that will transform the original environmental recording and reproduce it with a reduction of its natural complexity. This translational condition is not exclusive of music and its tools, it extends to a broad expanse of human activities. Ablinger poses the problem from the perspective of painting.

3.2.2.2. Cézanne, Photorealism and Music

With a background in the visual arts before devoting himself to composing with sound, Ablinger expresses a particular interest in Cézanne's notions on perception. In his lecture *Cézanne and Music* (2013), presented at the Historical and Contemporary Modes of Musical Listening Symposium in Graz, Austria, lies a connection between his understanding of Cézanne's notions of cubism and his idea for *Quadraturen IV*. Ablinger recalls that when Cézanne painted a line, the result was its deconstruction.

> If as an exercise we would subject ourselves to insistent and precise observation, we would recognize that a line is actually not a line—that it jumps here and there, that it is sometimes stronger and sometimes weaker, sometimes sharper and sometimes more blurred (…). Cézanne did not paint what he saw, he painted seeing. (Ablinger, 2014)

He was not painting reality: he was painting his perception of reality. This is a shared fundamental concept in *Quadraturen IV* in that it presents "the truth of the observation process" and derives from it an "aesthetic phenomenon." (Ablinger, Quadraturen IV, n.d.) The observation process is revealed in the music as the varying pulsed grid indicative of the various degrees of resolution we are able to attain through our perception and our technological tools—in the case of *Quadraturen IV*, spectral analysis.

Perception and representation of reality present a different kind of complexity in instrumental music—*Quadraturen IV*'s medium—than in painting. Painting translates the forms of environments of the real world in a way in which the spectator does not need any pointer or program note to realize that a reference to the real world is being made. On the other hand, when an ensemble of musicians is the medium through which a composer translates the sound of an environment of the real world into music, the resemblance of the music to the real-world environment is more complicated and, in most renditions, requires program notes for the spectator to be able to establish the relationship. In *Quadraturen IV*, this situation may be seen as an oversimplification of reality when it is perceived through the tools of music analysis, notation and instrumentation.

Translation and reproduction of the real-world objects has been a prominent practice in most arts throughout history. Painters have studied the anatomy of the human body, the color temperature in landscapes, perspective in facades and so on, to then reproduce the subjects of their perception through the language and tools of painting. Writers reproduce and recreate the real world through words that evoke images, emotions and situations often situated in time. With the exception of dance and instrumental music, there are several examples of other art forms in which the artist is an observer of a reality that is reproduced to a degree in which the spectator does not need previous explanation to know that what is being presented is a reproduction of a given reality (i.e., a person, an object of nature, etc.). In painting, the material for production is paint; in instrumental music, the material for production is the sound that the instrument makes. Paint, brushes and the various canvases available to the painter are able to present the qualities of the shapes, colors, texture, etc., of a given object that is being translated from the real world into the painting. On the other hand, the timbre, amplitude, rhythm, and frequency palettes proper of musical instruments present more difficulties to translate the qualities of a given sound in the world in a way in which it is clear to the spectator. If one takes into consideration the differences in affordances proper to drawing lines in painting versus making melodies in instrumental music, there is a striking difference that reveals how instrumental music has a significant limitation related to resolution.

We know that real world sounds most often exist as complex continua that by far transcend the resolution of the octave divided depending on the standard tuning in use. It is perhaps because of these limits in the affordances of the materials and techniques in instrumental music that representation of environmental sounds has not reached a reproduction that is precise enough that it does not need program notes. The only representational route taken by instrumental music to reproduce sounds of the world has involved metaphor—the presentation of sounds that possess similar characteristics to those of the original in pitch, duration, density, etc. This distinctive aspect of program music makes program notes essential to point the listener's imagination towards a given represented/reproduced subject. *Quadraturen IV* shows an intent to achieve a more precise representation of a complex acoustic reality. This music does not aim at a metaphorical representation of acoustic reality. Ablinger's use of the word "phonograph" as related to the word "photograph" is suggestive. While an "image" may imply various degrees of distance or proximity to reality, the word "photograph" arguably presents a more detailed translation of reality. The goal in the *Quadraturen* cycle is to grasp the phonographs' frequency, time and amplitude as closely as the media of translation (music technology, including notation and electronics) and reproduction (musical instruments and musicians) allow. In this sense, Ablinger establishes the connection between the time/frequency grid proper of spectral analysis and the gridding technique used by photorealist painters to achieve high degrees of resolution in the translation of the photograph to the painting. “The reproduction of ‘phonographs’ by instruments can be compared to photo-realist painting, or—what describes the technical aspect of the "Quadraturen" more precisely—with techniques in the graphic arts that use grids to transform photos into prints.” (Ablinger, Phonorealism, n.d.)

3.2.2.3. Reality through Music

Despite the point of departure of the phonograph and its translation into music, the ultimate objective of *Quadraturen IV* and the entirety of the *Quadraturen* cycle is not the literal reproduction of the phonograph itself, but the boundaries between “abstract musical structure” and “phonorealism”. In other words, Ablinger’s ultimate purpose is to explore the observation of “reality” through “music”. Throughout the cycle, reality is presented through different degrees of distance and proximity depending on the tools that are being used for translation and reproduction. The relationship between musical abstraction and the original recording—the resemblance between the music and the original—is less frequent in *Quadraturen IV* than in other pieces of the *Quadraturen* cycle in which the means of reproduction are electronically driven. In this sense, the analyses and translations of original recordings into instrumental music present less proximity to the originals than the translations into electronically manipulated instruments or media. This distance from the original is related to the affordances of instrumental performance. In this sense, Ablinger mentions the following:

> When using humanly played instruments (which is the case in *Qudraturen IV*) the grid has to be enlarged (slowed down) to remain playable—thus the result of the transformation is not so much a reproduction of the original but an approach to or a

situation of comparison between instrumental sounds and the original sound source." (Ablinger, Quadraturen IV, n.d.)

On the other hand, when reality is explored through the affordances of electronic music, the translations present a higher resolution—a grid of smaller squares and therefore a higher quantity of sonic information per second—and the reality is reproduced with a higher degree of proximity to the original. Such is the case in *Quadraturen III–Deus Cantando*, for computer-controlled piano and screened text. In this piece, the piano is reproducing the spectral analysis of the Declaration of the International Environmental Criminal Court, by Adolfo Pérez Esquivel and the XIV Dalai Lama. The similarities of the sound of the computer-controlled piano and the sound of human speech are striking. In contrast, the pieces in the *Quadraturen* cycle that involve orchestration and human performance present a lower resolution in order for the material to be performable. The result—which yields the important reflection on the limits of perception—is a transformed reality, simplified and reduced to comparatively larger squares on a grid, which is a condition of musical language itself. This distance from reality, due to the limits of human perception and of the tools of music as translators of objective parameters of environmental sounds, results in a unique and new sound world that is still, however, quantifiably—with some degree of objectivity—related to reality. The end result of the limitations of the translation is, as Ablinger states, an aesthetic phenomenon in itself.

3.2.3. Limits of perception and complementarity

As suggested above, *Quadraturen* may be seen as a study of varying modes of listening and of the perception of sound. With regards to perception, Humberto Maturana, biologist involved with studies in second order cybernetics, alerts us about epistemological and linguistic limits: "One can only say with a given language what a given language permits." (Maturana, 1980, p. xiii) Environments outside of musical thought, either natural or urban, possess a complexity that human perception and musical tools are able to express and reproduce only in a limited resolution. Through the tools of Digital Signal Processing, it is possible to reproduce the sound of any musical instrument through frequency modulation or additive synthesis. However, the reproduction will inevitably exclude minute aspects of the original sound and we will hear the gridded language of signal processing that is unable to reach the complex sound of the real instrument. Furthermore, if a single, discrete, musical instrument of either periodic or aperiodic sound presents such difficulties to be perceived (analyzed) and reproduced, the degree of complexity that non-musical sounds and entire environments—natural and urban—may present is far greater. The sound of busy intersection in a city, with a multiplicity of bodies of various materials generating vibrations expressed in periodic or a-periodic sounds, presents a complexity that exceeds our sonic perception, musical language and means of representation through instrumental sounds. In Maturana's terms, *Quadraturen IV* explores what musical language permits to say about an acoustic reality or environmental recording.

3.2.3.1. Autopoiesis

With regards to his initial observation on the limits of language, Maturana (1980) suggests that a new approach to cognition must be considered: we are required to "treat seriously the activity of the nervous system as determined by the nervous system itself, and not by the external world; thus the external world would only have a triggering role in the release of the internally-determined activity of the nervous system" (p. xv). This idea is the basis for the concept that he coined as *autopoiesis* (p. xxii), which refers to the self-organization of a living system. The principle of *autopoiesis* applies to the relation between the music of *Quadraturen IV* and the recording or phonograph from which it is derived. The nervous system described by Maturana is analogous to our musical tools, which in *Quadraturen* include spectral analysis, the composer's interpretation of it, and existing musical notation and orchestration resources. The external world is the recording, or phonograph, of Berlin. The activity of the nervous system is the musical translation of the external world—also a recording of Berlin. Such activity or musical translation becomes a new version of the external world and not its reproduction. An example in *Quadraturen IV* is the rhythmic resolution, which is mostly ostinato quarter and eighth notes in all parts, while the frequency resolution is semitones. This translational feature is determined by the tools of music and the composer perceiving a soundscape of Berlin, and not by the soundscape of Berlin itself. This idea may be examined through the lens of Maturana's observations on frog and pigeon vision: "perception should not be viewed as a grasping of an external reality, but rather as the specification of one, because no distinction was possible between perception and hallucination in the operation of the nervous system as a closed network." (p. xv)

3.2.3.2. Deficiency of attention

Another aspect that is relevant to *Quadraturen* is Jonathan Crary's (1999) idea regarding the "deficiency of attention" (Crary, 1999, p.1), which Ablinger expresses as: "to pay attention to one thing means withdrawing it from many other simultaneous things." (Ablinger, 2014, p.3) In *Quadraturen IV*, what Ablinger calls an "oversimplification of reality" may not only apply to the enlargement of the squares in the grid that defines the resolution of the translation, but also to the deficiency of attention that results from privileging only one aspect or a limited group of aspects of the phonograph and, at the same time, withdrawing the attention from many other qualities in the sound. For example, the technical affordances of the spectral analysis tools exclude the definition of discrete sound objects featured in the recording, i.e., the Doppler effect of a bus passing by. The spectral analysis focuses on the totality of the harmonic spectrum of the recording, withdrawing its focus from discrete events with their specific timbres and envelopes.

If, on another translation of *Quadraturen IV*, the attention was focused on the sound object of the bus passing by, then it would be withdrawn from the totality of the harmonic spectra of the general soundscape. This is of course a limit of our perceptual apparatus, and this idea is brilliantly instilled—perhaps even with a slight degree of irony—in the sound of *Quadraturen IV*. With regards to ways of perceiving paintings, Crary (1999) proposes "to construct some of the field of their *exterior*, to multiply the links to this exterior, 'to remain attentive to the plural', where 'everything signifies ceaselessly and several times'" (p. 9). This mode of perception as a spectator may as well shed light on how to perceive from the perspective of the translator of reality into art. In the particular case of translating environmental sound into music for performers, "to remain attentive to the plural" may be understood as including as many aspects of the recording that we may be overlooking due to the "paying attention" to a specific aspect of the recording. This notion of Crary points towards complementarity, which anthropologist and cyberneticist Gregory Bateson features as a chapter in *Mind and Nature.*

3.2.3.3. Complementarity

From an evolutionary standpoint, Gregory Bateson states the importance of combining pieces of information.

> The evolutionary process must depend upon such double increments of information. Every evolutionary step is an addition of information to an already existing system. Because this is so, the combinations, harmonies, and discords between successive pieces and layers of information will present many problems of survival and determine many directions of change. (Bateson, 1979, p. 19)

An approach to the translation of environmental sounds into music for performers that includes more than one method of analysis and notation would add additional information to the existing system. From a Batesonian perspective, this approach would contribute to the process of understanding of our environmental sounds and the tools that we use to study and reproduce them. The result of such combination of methods of translation of a phonograph into instrumental music could also be described using a metaphor with Bateson's description of binocular vision:

> The binocular image, which appears to be undivided, is in fact a synthesis of information from the left front in the right brain and a corresponding synthesis of material from the right front in the left brain. Later these two synthesized aggregates of information are themselves synthesized into a single subjective picture from which all traces of the vertical boundary have disappeared. (1979, p. 65)

This synthesis that takes place in binocular vision (described above) may be used as a metaphor to illustrate what would take place with two or more systems of translation of the same phonograph (soundscape) into instrumental music. Those systems, or notations, may be synthesized either by superimposition (simultaneously) or by collage (jumping from one system to the other in various orders and proportions). *Quadraturen IV* presents various sections in which there is a superimposition of time resolution expressed in a simultaneity of pulses (Figure 30). "From this elaborate arrangement, two sorts of advantage accrue. The seer is able to improve resolution at edges and contrasts; and better able to read when the print is small or the illumination poor." (Bateson, 1979, p. 65) This notion of aggregating information from various perceptual standpoints is explored in the compositional proposal below.

3.2.4. Conclusions: a compositional proposal

Spectrally and sonic data-based translation of environmental sounds into acoustic music is a compositional practice that is developing. Despite the time (1998) since it was composed and premiered, Peter Ablinger's *Quadraturen IV* stands as one of the relevant works in this still evolving method of music composition that translates environmental recordings into music for performers with an objective, quantifiable perspective through signal analysis. This work brings forth important considerations regarding the various potential and unique aesthetic phenomena that may result from the limits and different modes of perception, skills, technologies, and choices involved in the translation processes.

Through *Quadraturen*, Ablinger suggests that different musical tools and languages of observation and reproduction yield different, but complementary, versions of an acoustic reality. While one version may present a higher degree of resolution in some features of a given soundscape from the real world, another version may present a lower resolution in the same features and at the same time offer other valuable angles of that reality. As a result of the limitations and complementarity of perceptions, translations and reproductions, the idea of a superimposition and/or successive juxtapositions of different processes of translation of the same environmental soundscape is a promising path for further work based on environmental soundscape translation.

We have seen in *Quadraturen IV* how there is a systematic decrease and increase in resolutions of time that result in superimposed or juxtaposed tempi. This notion of simultaneity of angles of perception may be the antecedent of a simultaneity of musical systems and versions that originate from the same source but that are a result of different processes. This artistic and conceptual approach to the translation of environmental sounds into instrumental music can potentially offer a richer and more complete understanding of the sounds of our world. In his proposal of *multiple versions of the world*, Gregory Bateson states that "the difference between the information provided by the one retina and that provided by the other is itself information of a *different logical type*. From this new sort of information, the seer adds an extra *dimension* to seeing" (Bateson, 1979, p. 65). Similarly, with the imprint of the composer's tools, different processes of translation and instrumentation of environmental recordings add extra dimensions to listening, in which objective observation of the structure of the sound is fused with the subjective characteristics of the translator. As Ablinger states in his subtitle for *Quadraturen IV,* such extra dimension to listening is in fact a *self-portrait* with the acoustic phonograph, a fusion between the listener and the sound.

IV. Environmental Sonic Translation Methodologies: from the Model to the Composition

This chapter discusses technical and aesthetic criteria involved in my own compositional work. In the main sections of this chapter (4.1 and 4.2), I describe the methodology and compositional aims of four compositions that I have created from the traits of geophonic and biophonic soundscapes. The soundscapes from which the compositions are drawn may be roughly characterized by either of two general spectromorphological[4] characterstics of sound—complex and pitched—described below. These categories refer to whether a sound model is pitch-defined or noise-defined.[5] The spectromorphological traits of a sound model define the resulting analysis, compositional concept, notation, and instrumentation (either in terms of instrument choices or instrumental techniques).

In my work, a higher degree of noise in the sound model often yields a higher degree of indeterminacy in the transcription notation. This indeterminacy is paired with a high harmonic density, often presented as clusters. While noise is characterized by a saturation of partials in the entire frequency spectrum, clusters consist of a saturation in the entire pitch spectrum. On the other hand, a higher degree of pitch definition contained in the sound model derives in determinacy and single pitch specificity. As mentioned above, whether a sound is noised or pitched is a defining factor for the pre-composition, analysis and general concept of the piece. Techniques discussed in Chapter II, such as the manual-aural transcription, partial tracking analysis, and spectral harmonic analysis, are chosen according to such spectromorphological features of the material. As an example, manual-aural transcription, frequently done through listening to a recording of the model and manually notating its musical features, would be a humorously unfeasible approach to a sound model characterized by broad band noise such as a waterfall. For that case, a technique such as partial tracking yields results and possibilities appropriate to the varying complexities of frequency organization characteristic of broad band noise.

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[4] Denis Smalley (1997) coined the term "spectromorphology" to represent the components of the sound spectrum –the sound material and the pitch domain– and their evolution over time –the collective morphology.

[5] Naturally, there are sound models that present a mix of pitch and noise. These can be placed in either the Pitched or Complex category, depending on the pitch–noise ratio.

The sound models, corresponding pieces and analyses included in this chapter are placed within a continuum between the two aforementioned categories: pitched and complex. As is the case with most environmental soundscapes—very rarely do we find a purely pitched or entirely complex soundscape—, there is some overlap between all the pitched and complex soundscapes here presented. A complex soundscape may have some pitched features; a pitched soundscape may present some complex features.

Each category-section (Complex soundscapes / Pitched soundscapes) presents general procedures that I have designed for the analysis and musical translation of various pitched and complex sound models and their music translations. More specific approaches and techniques to visualize, transcribe, analyze, notate and organize the soundscape-derived sound models are described, addressing the affordances and musical potential of each sound model. The chapter presents selected examples of the musically notated scores for performance[6].

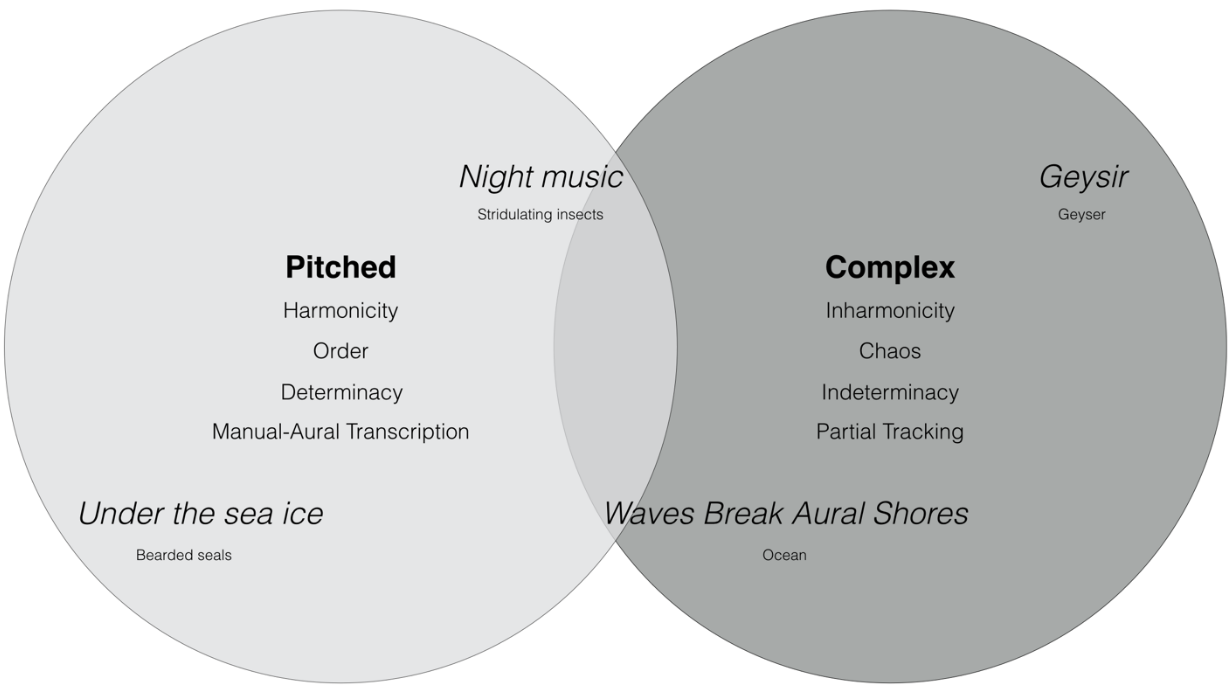

Figure 35. Diagram of featured works within the Pitched and Complex soundscape categories

[6] The full scores are presented in the Appendix

4.1. Complex Soundscapes: *Geysir / Waves Break Aural Shores*

The sounds of geological phenomena are generally noisy. Wind, glaciers, oceans, streams, and other geological sounds present a vast content of frequencies that often obscure individual pitches or groups of pitches. However, noise varies from sound to sound in terms of its parameters, of which my research and compositional methods have mainly focused on: 1. Frequency band predominance and 2. Amplitude patterns. Variances on these parameters contribute to the signatures that makes specific noise sounds unique. Such parameters also provide rich models for both formal and harmonic dimensions in sound model-based compositions. This section explains such procedures, present in *Geysir*, for seven pianists and electronics, and *Waves Break Aural Shores*, for saxophone quartet and electronics.

4.1.1. General analytic procedures

4.1.1.1. Partial Tracking, Frequency Region Segmentation and Loudest Frequencies Selection

Using SPEAR software, the audio recording is re-synthesized in order to manipulate, organize, and calculate the predominance of the frequency content (Figure 36). The FFT window size setting for the re-synthesis is based on the formula of a "duration of four to five times the period of the lowest frequency difference between the sinusoidal components of the sound"[7] (McAdams, et al., 2004, p. 175). Once re-synthesized, the audio is segmented in a given number of frequency bands, depending on the compositional aims of the piece. Each of these frequency bands becomes an independent file (Figure 37). The partials with amplitudes under a threshold based on the general waveform of the given sound model (i.e., -45 dB) are eliminated, remaining only the loudest ones (Figure 38).

[7] In *Waves Break Aural Shores*, FFT size variations are used in order to generate contrasting and complementary analyses and material.

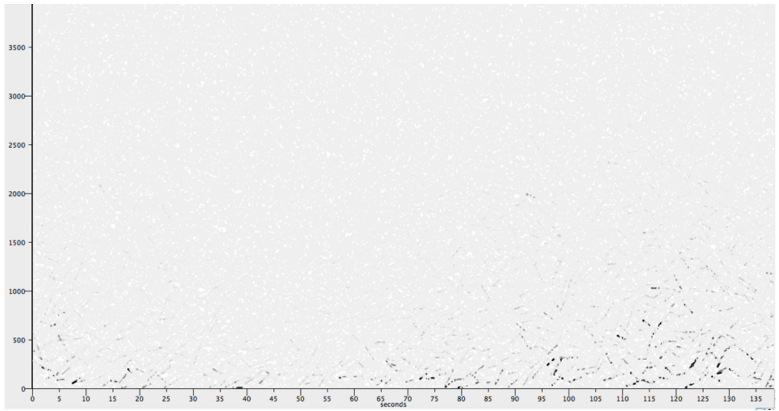

Figure 36. Entirety of re-synthesized partials (frequencies) of a complex soundscape

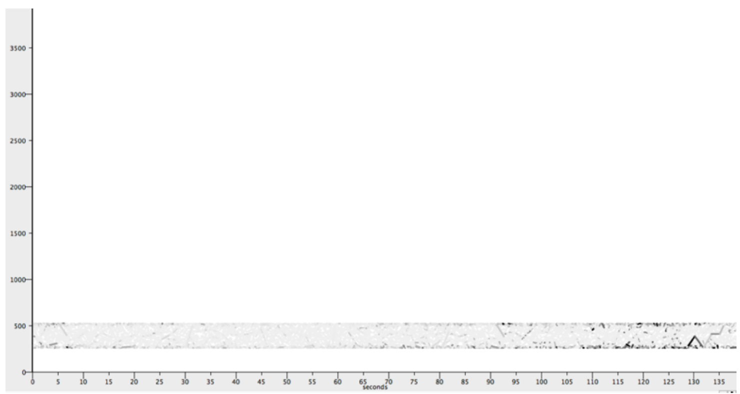

Figure 37. Segmented region (262-523 Hz)

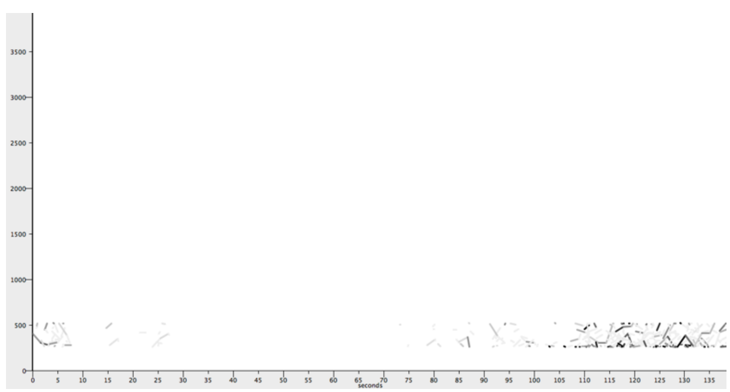

Figure 38. Segmented region after elimination of partials -25 dB

Using Max MSP or Open Music, the remaining partials, recorded in .sdif (Sound Description Interchange Format), are then converted into music notation consisting mainly of pitch and duration. In both software applications, quantization of the resulting rhythms and micro-tonal tunings of the sound model transforms the content into desired rhythms adequate for pulse/time reference and tuning systems.[8]

The quantized audio data is translated into Music Exchangeable Markup Language format (.xml), which makes the contents compatible with data applications and music engraving software such as Sibelius or Finale.

4.1.1.2. Amplitude Analysis

Due to the lack of discernible harmonic or motivic content due to frequency saturation in complex sound models and soundscapes, amplitude trajectories and curves become an important analytic attribute. The parameter of amplitude reveals clear contours that allow soundscapes characterized by noise to become sound models for any kind of compositional project.

The main procedures implemented for the amplitude analysis of complex soundscapes have been performed manually. While music information retrieval (MIR) tools could be useful for automating the task, the manual approach leads to more listening exposure to the sound model as well as a creative malleability allowed by the medium. Such advantages are crucial for the composition phase after the sound model analysis.

Form and melodic material can be derived from the amplitude analysis using the following analytic procedures.

4.1.1.2.1. *Amplitude-derived Form*

Amplitude analysis of the sound model defines the form in the two pieces composed using complex soundscapes—*Geysir* and *Waves Break Aural Shores*. In *Geysir*, density in terms of number of active electronics tracks (1 through 7) is defined by the amplitude contours of the general waveform (see 4.1.2.2.6). In *Waves Break Aural Shores*, the types of material, techniques, and dynamics are mapped to sections defined by eight amplitude categories (1–

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[8] In *Geysir,* equal-tempered tunings were chosen due to the fact that the music resulting from this analysis would be written for seven pianos. Quantization choices will vary depending on the affordances of the instruments that will perform. String instruments and some wind instruments can perform micro-tonal divisions up to $^1/_8$ of a tone, as opposed to the piano, tuned to equal ½ of a tone.

lowest amplitude; 8–highest amplitude). Each amplitude category-derived section corresponds to a breaking of a wave in the field recording. Thus, each wave cycle is a section in the piece. These sections are categorized based on eight amplitude level ranges derived from the 48 waves breaking in the recording.

The two main types of amplitude analysis are shown below (Figures 39 and 40): 1. Amplitude contours; 2. Peak amplitude categories.

1. Amplitude contours. The contours of the wave form of the geyser sound model are drawn manually, following its general shape. Stratification of the dynamics is assigned arbitrarily, based on my aural experience the sound model *in-situ* (while recording the sound).

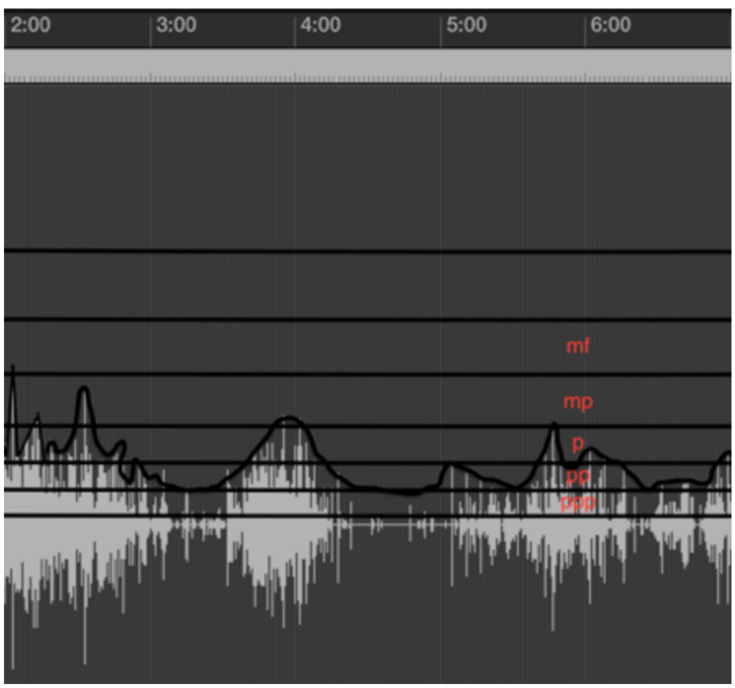


Figure 39. Fragment of the geyser recording amplitude contours

2. Peak amplitude categories. Markers were used to divide each of the 48 breaking waves in the field recording of the ocean:

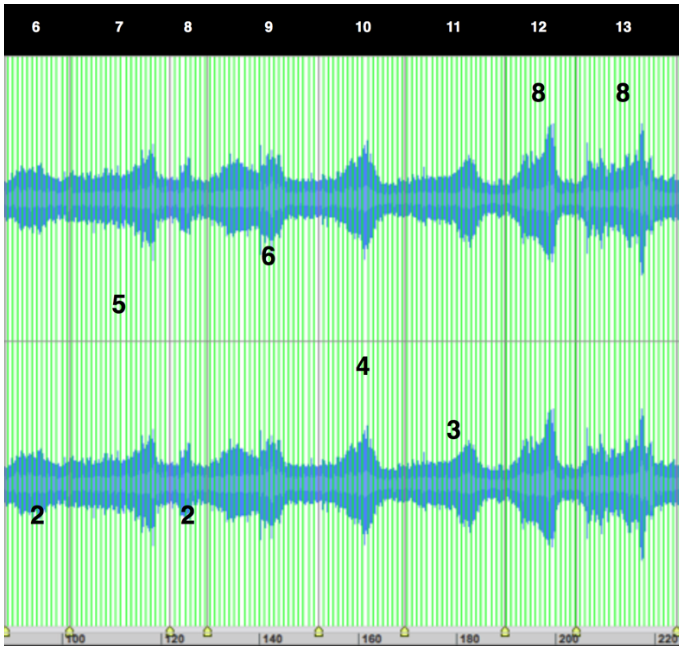


Figure 40. Fragment of the ocean recording waveform with markers
(waves 6-13, as indicated by the numbers at the top of the figure.

In each of the 48 sections, the peak amplitude was detected using the peak level display in the DAW. The 48 peak levels were distributed into eight categories (Figure 41):

8

Section/Wave	Time code	Peak amplitude LP	Peak amplitude HP
12	03:09.8	-0.5	-1.2
13	03:24.2	0	-2.9
39	08:16.8	0	-2.2

7

Section/Wave	Time code	Peak amplitude LP	Peak amplitude HP
16	04:06.0	-1.6	-6.7
18	04:30.1	-1.2	-1.2
45	09:23.6	-1.3	-0.1

6

Section/Wave	Time code	Peak amplitude LP	Peak amplitude HP
9	02:09.5	-1.9	-5
34	07:30.0	-2.2	-0.6
49	10:11.6	-2.3	-4.3

5

Section/Wave	Time code	Peak amplitude LP	Peak amplitude HP
7	01:41.5	-2.9	-4.9
14	03:44.6	-2.8	-8.8
30	07:03.8	-2.7	-0.1
32	07:15.6	-2.5	-0.1

4

Section/Wave	Time code	Peak amplitude LP	Peak amplitude HP
10	02:31.9	-3.1	-6.1
25	06:15.1	-3.3	-8.8
28	06:46.2	-3	-5.2
31	07:07.3	-3.1	-3.1
33	07:22.9	-3.6	-4.2
38	08:05.8	-3.6	-3.7
40	08:24.2	-3.1	-6.1

3

Section/Wave	Time code	Peak amplitude LP	Peak amplitude HP
11	02:49.3	-4.9	-9.4
17	04:18.0	-4.8	-8.3
20	04:55.0	-4.4	-11
21	05:03.0	-4.1	-5.4
23	05:23.8	-4.9	-4.1
27	06:36.3	-5	-5.2
29	06:58.2	-4.1	-2.6
35	07:35.8	-4.8	-5.1
37	07:49.8	-4.9	-5.7
43	09:03.0	-5	-9.2
48	09:51.3	-5	-3.6

2

Section/Wave	Time code	Peak amplitude LP	Peak amplitude HP
4	00:55.3	-5.4	-15
5	01:12.2	-5.9	-8
6	01:28.5	-6.2	-12
8	02:01.7	-5.5	-4.5
22	05:14.0	-6.3	-6.2
24	05:57.1	-5.4	-6.4
26	06:27.0	-6.1	-6.8
36	07:45.3	-5.9	-2.8
41	08:41.8	-5.9	-6.1
42	08:51.0	-5.5	-12
46	09:33.4	-6.1	-2.6
47	09:40.0	-5.5	-5.2

1

Section/Wave	Time code	Peak amplitude LP	Peak amplitude HP
1	00:00.0	-11	-22
2	00:22.6	-7.9	-20
3	00:42.0	-7.1	-18
15	03:58.7	-6.8	-11
19	04:44.9	-7	-13
44	09:14.6	-6.7	-13

Figure 41. Ocean recording's peak amplitude categories

4.1.1.2.2. *Varying-amplitude-threshold-analysis-derived melodic material*

Another use of the amplitude parameter in the analysis of complex soundscapes is the generation of multiple motivic versions of the same sound model. In the selection of loudest frequencies of a re-synthesized audio file (see 4.1.1.1.), varying amplitude thresholds of a single sound model's re-syntheses generate various layers of motivic material. These may be used as different voices in a polyphony consisting of varying analytical perspectives of the model (see 4.1.3.1.4.). In SPEAR, a number of re-syntheses may be derived from the same sound model under varying minimum amplitude thresholds. For example, -19dB, -20dB, -21dB, and -22 dB, respectively. The results are four re-synthesizes, each with streams of partials from a lower amplitude threshold of -22 dB to a higher amplitude threshold of -19 dB (Figures 42–45). Each of these .sdif files are converted into music notation in OpenMusic. This notation may then be organized as material in the score.

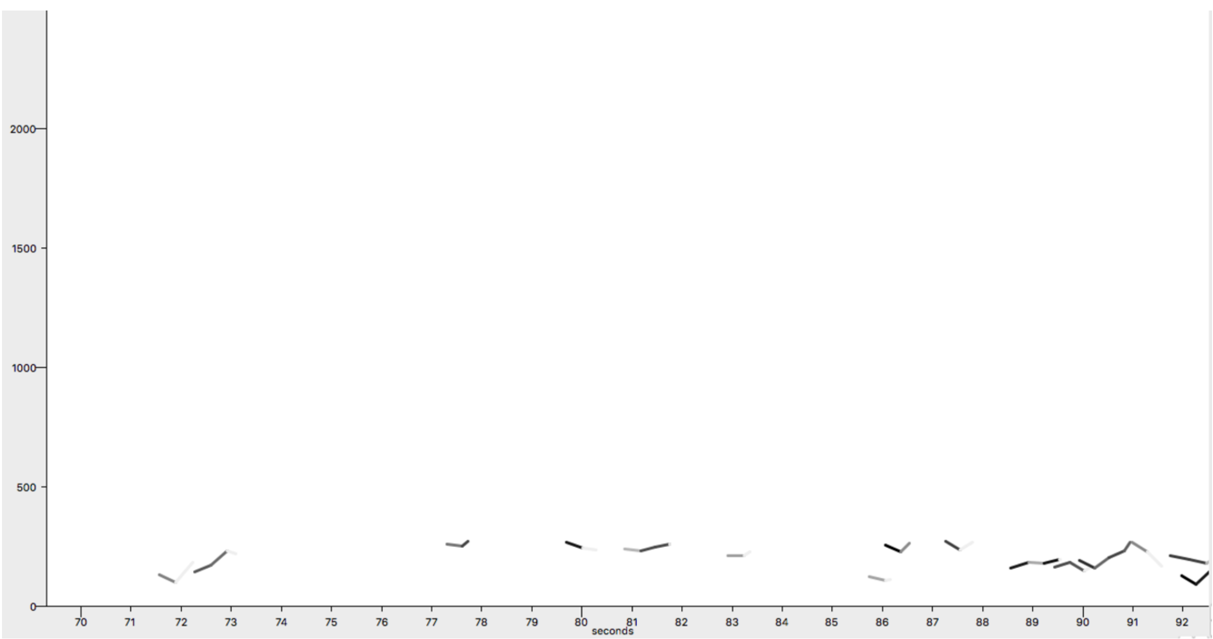


Figure 42. Four re-syntheses of the same ocean audio: minimum amplitude threshold: –22dB

Figure 43. Four re-syntheses of the same ocean audio: minimum amplitude threshold: –21dB

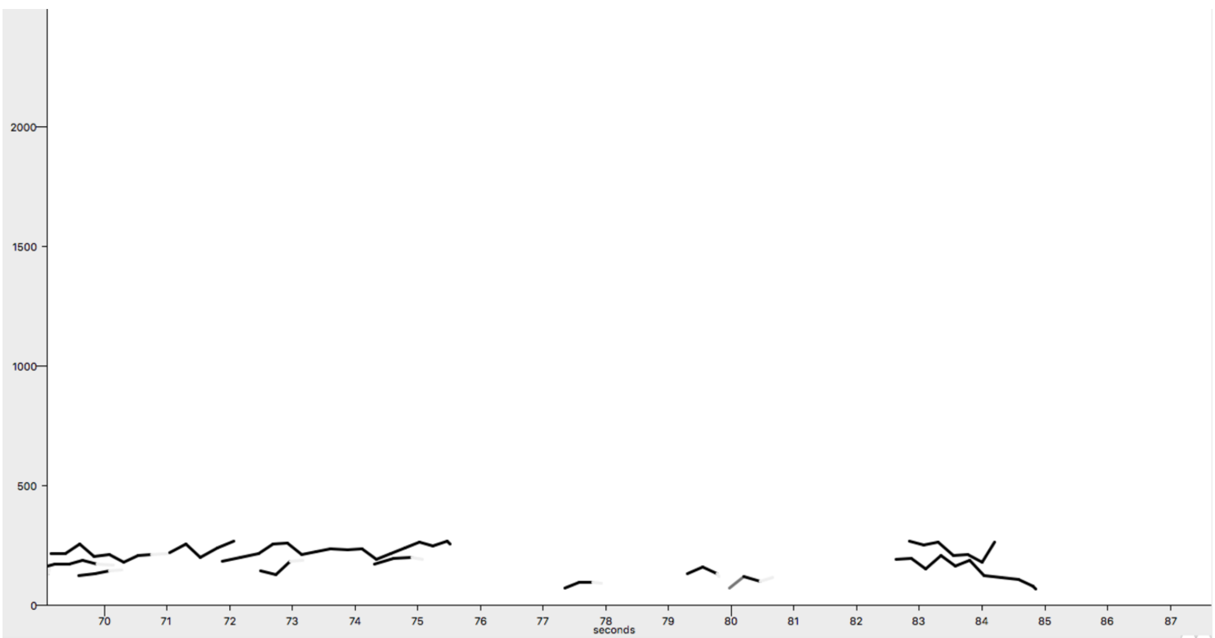


Figure 44. Four re-syntheses of the same ocean audio: minimum amplitude threshold: –20dB

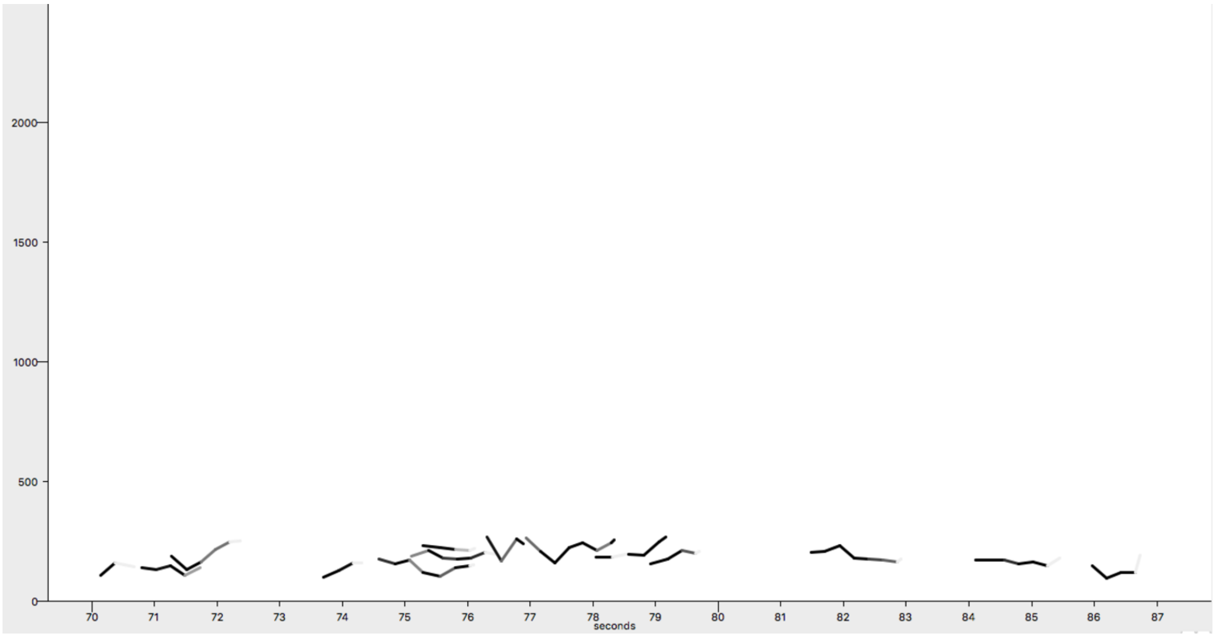


Figure 45. Four re-syntheses of the same ocean audio: minimum amplitude threshold: –19dB

These general analytic procedures are used in the pieces shown below: *Geysir*, for 7 pianists and spatialized electronics, and *Waves Break Aural Shores,* for saxophone quartet and electronics.

4.1.2. *Geysir*: pitch class predominance/amplitude analysis and musical translation of geological noise

In this piece, the sound of one the geysers in the Geysir system of the Haukadalur valley, 180 miles Northeast of Reykjavik, Iceland, is recorded and analyzed in multiple time segments, each with its own pitch-class set predominance. The analysis is further adapted into a piece for seven spatialized pianists and electronics, *Geysir*, which features the amplitude and predominant pitch class fluctuations throughout the geyser sample.

This study analyzes 11 minutes of recording using a suite of software for the identification of amplitude and frequency (pitch) content. The pitches presented in this analysis are quantized—simplified—to equal temperament (C, C#, D, D#, etc.) in order to be adapted to the piano's tuning affordances.

4.1.2.1. Field Recording

The recording was made with two DPA 4060 omnidirectional microphones with a frequency response peaking at 3.5 dB at 10.3 kHz.

4.1.2.2. Compositional / Analytical Approach

The analysis was defined by the compositional approach to the geyser sound model. The music derived from it aimed to embody the complexity of noise, characteristic of powerful geophonic sounds such as a geyser. However, such complexity needed simplification in order to reveal salient features of the sound model and make the music not exceedingly difficult to perform.

Re-synthesis derived partial tracking—explained below—was used for the initial stage. The re-synthesis process included octave segmentation and the deletion of partials below an amplitude threshold. Notation prototypes derived from the re-synthesis were generated in Open Music, yielding quantized material of a high degree of complexity (Figure 46).

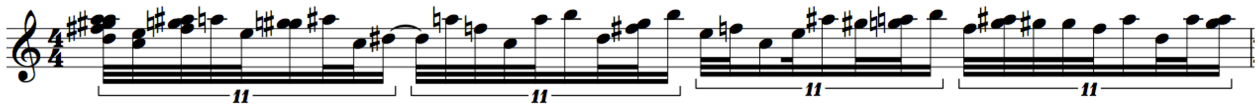


Figure 46. Staff 3 (C5-B5), 00:00-00:03, determinate partial tracking derived notation

Considering the sound model's constant density and saturation from beginning to end, the performers would have had to sustain the rhythmic characteristics of this measure throughout the ~10-minute piece. From a perceptual perspective, the brevity of the rhythmic durations paired to the saturation of the pitch sets is heard as a random mass of sound instead of auditory streams. The performance challenges of the material in addition to its aleatoric sound led to consider indeterminate procedures for performance and rehearsal time economy. Such procedures required the performers to generate the rhythmic material in a guided-improvisatory manner, with instructions regarding pitch contents, target rhythmic densities, and dynamics. This idea led to a further distillation of the sound model: the calculation of pitch predominance at any desired point in time, organizing the pitches in three categories: high, medium and low predominance. The calculation was based on the total pitch count and durations per groups of measures, which provided the target rhythmic densities for each section. The pitch categories were assigned to specific note-heads (explained in 4.1.2.3).

Figure 47. Staff 3 (C5-B5), 00:00-00:03, indeterminate pitch-predominance notation

While the first example (Figure 46) expresses the first four seconds of the geyser in the frequency range between 523 and 1047 Hz (C5-B5) with determinate notation, the second example (Figure 47) expresses the abstraction of the first 16 seconds of the same frequency range with indeterminate notation. The differences between the sound of the two versions were strikingly low, the sonic signature of the salient predominant pitches and the rhythmic density of the section were preserved, and the material was made more accessible for performers.

A significant byproduct of the pitch-predominance analysis and resulting indeterminate notation was the increased potential of the material to be embodied by the performers. Their agency with the indeterminate material establishes a closer connection to the sound model, and therefore to the geyser.

The following sub-sections will explain the process for the pitch, rhythm and dynamics analysis and notation of the geyser's sound model.

4.1.2.2.1. *Frequency region segmentation and partial tracking*

Using SPEAR software, the audio recording was re-synthesized in order to manipulate, organize, and calculate the predominance of the frequency content (Figure 48). Once re-synthesized, the audio was segmented in seven regions, from high to low, equivalent to a piano's seven complete octaves, from the lowest (C1) to the highest (C7). Each of these regions became an independent file (Figure 49). The partials with the amplitudes under -45 dB (the quietest) were eliminated, remaining only the loudest ones (Figure 50).

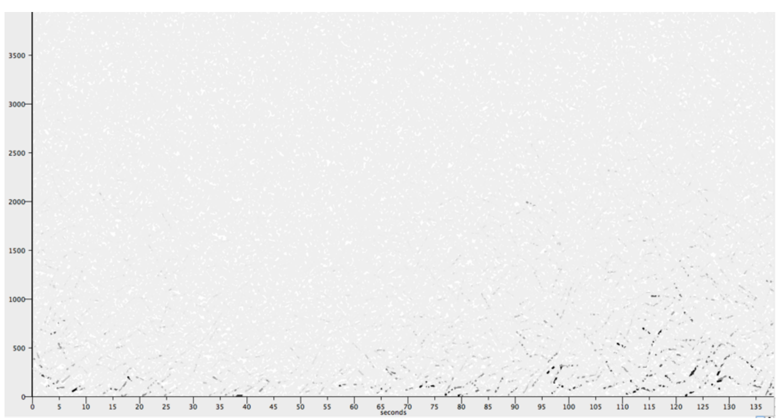


Figure 48. Geyser analysis, entirety of re-synthesized partials until 2:15

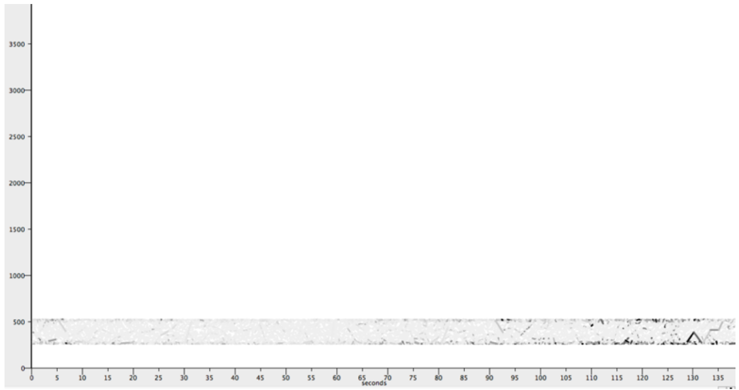


Figure 49. Segmented region (262-523 Hz)

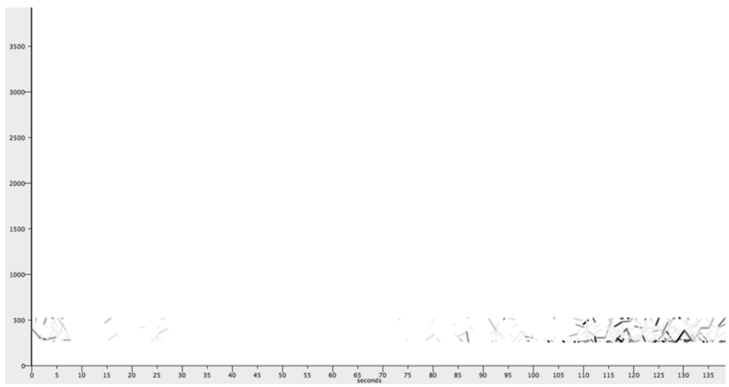


Figure 50. Segmented region after elimination of partials -25 dB

The sound data was converted from the SPEAR Sound Description Interchange Format (.sdif) into a text file (.txt) with IRCAM's Orchidée computer aided orchestration software (Carpentier & Tardieu, 2008). Using Max MSP's Bach object library (Agostini & Ghisi, 2015), the re-synthesized audio data encoded in the .txt file was then converted into music notation via partial tracking (Figure 51). A tool was generated for quantizing the complex rhythms and micro-tonal tunings of the geyser into simple rhythms adequate for pulse/time reference and the chromatic equal-tempered tuning system.[9] The resulting quantized audio data was translated into an .xml file, making the contents compatible with data applications and music engraving software such as Sibelius or Finale.

[9] Equal-tempered tunings were chosen due to the fact that the music resulting from this analysis would be written for seven pianos. Quantization choices will vary depending on the affordances of the instruments that will perform. String instruments and some wind instruments can perform micro-tonal divisions up to $^{1}/_{8}$ of a tone, as opposed to the piano, tuned to equal ½ of a tone.

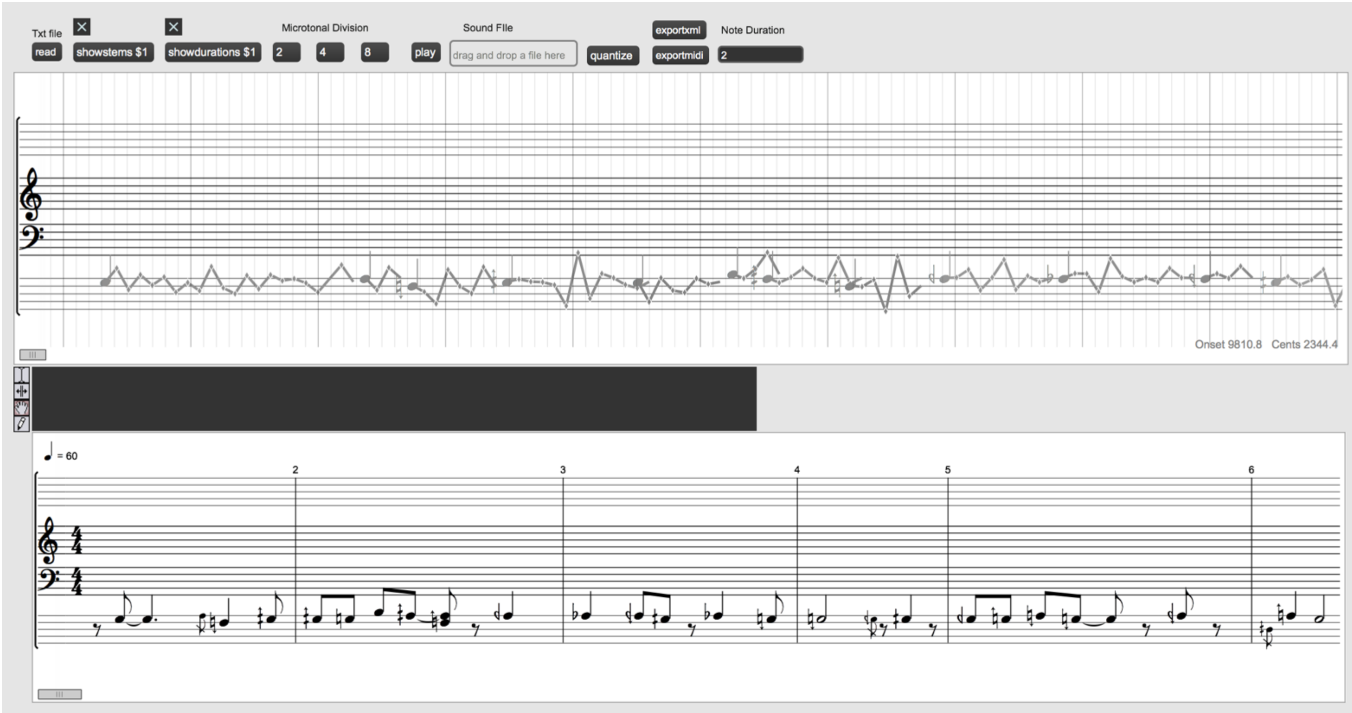


Figure 51. Max MSP patch, including the bach.roll, bach.score, quantization, and .xml conversion tools (Tfirn, n.d.)

4.1.2.2.2. Pitch class generation by predominance

The last phase of the pitch analysis was the classification of the geyser's pitch classes derived from the previous processes. Using the BaseX database engine, all the pitches in the .xml file were organized by predominance and octave under the criteria of onset count and duration. A custom XQuery script (Version 3.1, World Wide Web Consortium, 2017) grouped the pitch classes and added the total time of their onsets within a specific time segment (i.e., 16 onsets with a total of 20 seconds within a 25 second segment). Lastly, a list in descending order from most prominent to least prominent pitches was generated for time intervals varying between 12 and 24 seconds. The lists derived from this analysis were then adapted entirely into the musical score for *Geysir*, for seven spatialized pianists and electronics.

4.1.2.2.3. Pitch-class predominance analysis key

The information provided in this section is a description of each of the elements of the output from the XQuery script, which contains the distilled data used to inform the final scoring (Figure 52).

```
<group staff=”1” measures=”1,2,3,4”>
 <pitch value=”C#” duration=”1920” count=”15” rank3=”high ●” rank4=”high ●”/>
 <pitch value=”C” duration=”1408” count=”11” rank3=”high ●” rank4=”medium ◇”/>
 <pitch value=”Eb” duration=”1280 count=”10” rank3=”medium ◇” rank4=”medium ◇”/>
 <pitch value=”D” duration=”1280” count=”10” rank3=”medium ◇” rank4=”medium ◇”/>
 <pitch value=”F” duration=”896” count=”7” rank3=”medium ◇” rank4=”low ╳”/>
 <pitch value=”E” duration=”256” count=”2” rank3=”low ╳” rank4=”ruled out Ø”/>
```

Figure 52. Pitch predominance data for staff 1 at 0:00-0:20

Octave segmentation. The staves in the listing were numbered based on the initial octave segmentation performed in SPEAR.

- staff 1: C7 (piano 7) 2093-4186 Hz
- staff 2: C6 (piano 6) 1047-2093 Hz
- staff 3: C5 (piano 5) 523-1047 Hz
- staff 4: C4 (piano 4) 262-523 Hz
- staff 5: C3 (piano 3) 131-262 Hz
- staff 6: C2 (piano 2) 65-131 Hz
- staff 7: C1 (piano 1) 33-65 Hz

Pitch categories by predominance. The data in this document displays the pitches with highest predominance in each octave, from highest to lowest, within the noise of the geyser. These pitches are labeled in either of three predominance categories: high (●), medium (◇), low (╳), and/or ruled out (⦸). These encodings are kept for the performers in the musical score.

Time. Time is presented in measures. Each measure is 4 seconds long, and the listing shows calculations over grouped measures. For example, in the first segment of the analysis (i.e., <group staff=”1” measures=”1,2,3,4”>), each measure is 4 seconds long, so that the total time of measures 1, 2, 3, and 4 is 16 seconds. The temporal location of a measure is one less than the measure number multiplied by 4. For example, the time location of measure 30 is second 116, or time cue 01:56 (i.e., the first measure in `<group staff="1" measures="30,31,32,33,34">`).

Syntax. Each syntactic attribute in the document represents a specific feature used in the final scoring.

Feature	Description
Group staff	The octave analyzed (highest octave is group staff "1")
Measures	The total amount of measures in the segment analyzed
Pitch value	The pitch equivalence of the partial's frequency (e.g., 2218 HZ = C#)
Duration	Total duration of the partial's occurrences in the segment. The duration is displayed in milliseconds (1,920 milliseconds = 1.9 seconds)
Count	The number of iterations of the given pitch or frequency in the segment
Rank	The assessed predominance (low, medium, high). The "rank3" attribute is calculated from more pitches than "rank4", which filters out some pitches based on low counts/durations.

Figure 53. Elements of the pitch predominance analysis

4.1.2.2.4. Pitch-class Predominance Analysis in Music Notation

The examples in this section (Figures 54–60) feature the music notation and pitch content of selected measures of each segmented octave of the geyser recording, from the highest (staff 1) to the lowest (staff 7). The main elements from the pitch-class predominance analysis that are represented in music notation are: 1) The register, according to the octave categories shown above (Staff=1–C7 to Staff=7–C1); 2) The note-heads, which maintain the symbols used in the pitch-class predominance analysis.

The sound examples present the pitch collections in the following order:

1) as the displayed pitch collection in ascending order (as a scale), mapping loudness to predominance

2) as a cluster with all the notes sounding simultaneously at equal intensities

3) as a succession of chords featuring high predominance, medium predominance and low predominance pitches in succession

4) as a fragment of the actual performed part in the score

“Staff=1” | 2093-4186 Hz | 02:32 in recording and composition

```
<group staff="1" measures="38,39,40,41">
  <pitch value="C#" duration="8192" count="64" rank3="high ●" rank4="high ●"/>
  <pitch value="Eb" duration="4608" count="36" rank3="medium ◇" rank4="medium ◇"/>
  <pitch value="D" duration="3968" count="31" rank3="medium ◇" rank4="low ╳"/>
  <pitch value="C" duration="3712" count="29" rank3="medium ◇" rank4="low ╳"/>
  <pitch value="E" duration="2304" count="18" rank3="low ╳" rank4="low ╳"/>
  <pitch value="F" duration="896" count="7" rank3="low ╳" rank4="ruled out ⌀"/>
  <pitch value="F#" duration="896" count="7" rank3="low ╳" rank4="ruled out ⌀"/>
  <pitch value="Ab" duration="384" count="3" rank3="low ╳" rank4="ruled out ⌀"/>
  <pitch value="A" duration="128" count="1" rank3="low ╳" rank4="ruled out ⌀"/>
  <pitch value="G" duration="128" count="1" rank3="low ╳" rank4="ruled out ⌀"/>
```

Figure 54. Staff 1 notation

“Staff=2” | 1047-2093 Hz | 03:44 in recording and composition

```
<group staff="2" measures="57,58,59,60,61,62">
  <pitch value="D" duration="21632" count="169" rank3="high ●" rank4="high ●"/>
  <pitch value="F#" duration="19072" count="149" rank3="high ●" rank4="high ●"/>
  <pitch value="E" duration="13696" count="107" rank3="medium ◇" rank4="medium ◇"/>
  <pitch value="C#" duration="11776" count="92" rank3="medium ◇" rank4="medium ◇"/>
  <pitch value="F" duration="11776" count="92" rank3="medium ◇" rank4="medium ◇"/>
  <pitch value="G" duration="11136" count="87" rank3="medium ◇" rank4="low ╳"/>
  <pitch value="Bb" duration="9856" count="77" rank3="medium ◇" rank4="low ╳"/>
  <pitch value="Eb" duration="9472" count="74" rank3="medium ◇" rank4="low ╳"/>
  <pitch value="Ab" duration="8960" count="70" rank3="medium ◇" rank4="low ╳"/>
  <pitch value="A" duration="8320" count="65" rank3="medium ◇" rank4="low ╳"/>
  <pitch value="B" duration="5504" count="43" rank3="low ╳" rank4="ruled out ⌀"/>
  <pitch value="C" duration="896" count="7" rank3="low ╳" rank4="ruled out ⌀"/>
```

Figure 55. Staff 2 notation

“Staff=3” | 523-1047 Hz | 01:32 in recording and composition

```
<group staff="3" measures="23,24,25">
  <pitch value="Bb" duration="10240" count="80" rank3="high ●" rank4="high ●"/>
  <pitch value="E" duration="7936" count="62" rank3="high ●" rank4="high ●"/>
  <pitch value="Ab" duration="7168" count="56" rank3="high ●" rank4="medium ◇"/>
  <pitch value="B" duration="5632" count="44" rank3="medium ◇" rank4="medium ◇"/>
  <pitch value="F#" duration="5248" count="41" rank3="medium ◇" rank4="low ╳"/>
  <pitch value="F" duration="4736" count="37" rank3="medium ◇" rank4="low ╳"/>
  <pitch value="Eb" duration="4096" count="32" rank3="medium ◇" rank4="low ╳"/>
  <pitch value="D" duration="3968" count="31" rank3="medium ◇" rank4="low ╳"/>
  <pitch value="A" duration="3840" count="30" rank3="low ╳" rank4="low ╳"/>
  <pitch value="G" duration="3072" count="24" rank3="low ╳" rank4="ruled out ⌀"/>
  <pitch value="C#" duration="1920" count="15" rank3="low ╳" rank4="ruled out ⌀"/>
  <pitch value="C" duration="768" count="6" rank3="low ╳" rank4="ruled out ⌀"/>
```

Figure 56. Staff 3 notation

“Staff=4” | 262-523 Hz | 05:28 in recording and composition

```
<group staff="4" measures="82,83,84,85">
  <pitch value="Eb" duration="3584" count="28" rank3="high ●" rank4="high ●"/>
  <pitch value="A" duration="3712" count="28" rank3="high ●" rank4="high ●"/>
  <pitch value="E" duration="3584" count="27" rank3="high ●" rank4="high ●"/>
  <pitch value="F#" duration="3200" count="25" rank3="high ●" rank4="high ●"/>
  <pitch value="G" duration="2944" count="22" rank3="high ●" rank4="medium ◇"/>
  <pitch value="Bb" duration="2944" count="22" rank3="high ●" rank4="medium ◇"/>
  <pitch value="Ab" duration="2432" count="19" rank3="medium ◇" rank4="medium ◇"/>
  <pitch value="B" duration="2176" count="16" rank3="medium ◇" rank4="low ╳"/>
  <pitch value="F" duration="1152" count="9" rank3="low ╳" rank4="ruled out ⌀"/>
  <pitch value="D" duration="896" count="7" rank3="low ╳" rank4="ruled out ⌀"/>
  <pitch value="C#" duration="1024" count="7" rank3="low ╳" rank4="ruled out ⌀"/>
```

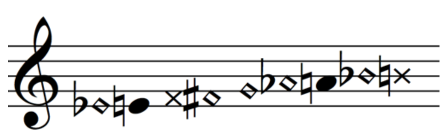


Figure 57. Staff 4 notation

“Staff=5” | 131-262 Hz | 01:32 in recording and composition

```
<group staff="5" measures="21,22,23,24">
  <pitch value="F#" duration="8320" count="64" rank3="high ●" rank4="high ●"/>
  <pitch value="Ab" duration="2816" count="21" rank3="low ╳" rank4="low ╳"/>
  <pitch value="G" duration="2560" count="20" rank3="low ╳" rank4="low ╳"/>
  <pitch value="E" duration="2176" count="17" rank3="low ╳" rank4="low ╳"/>
  <pitch value="F" duration="1408" count="10" rank3="low ╳" rank4="ruled out ⌀"/>
  <pitch value="B" duration="1152" count="9" rank3="low ╳" rank4="ruled out ⌀"/>
  <pitch value="C" duration="896" count="7" rank3="low ╳" rank4="ruled out ⌀"/>
  <pitch value="Bb" duration="640" count="5" rank3="low ╳" rank4="ruled out ⌀"/>
  <pitch value="A" duration="384" count="3" rank3="low ╳" rank4="ruled out ⌀"/>
  <pitch value="D" duration="128" count="1" rank3="low ╳" rank4="ruled out ⌀"/>
  <pitch value="C#" duration="128" count="1" rank3="low ╳" rank4="ruled out ⌀"/>
```

Figure 58. Staff 5 notation

“Staff=6” | 65-131 Hz | 05:40 in recording and composition

```
<group staff="6" measures="86,87,88,89">
  <pitch value="G" duration="6016" count="44" rank3="high ●" rank4="high ●"/>
  <pitch value="D" duration="768" count="6" rank3="low ╳" rank4="ruled out ⌀"/>
  <pitch value="F#" duration="768" count="5" rank3="low ╳" rank4="ruled out ⌀"/>
  <pitch value="Ab" duration="512" count="4" rank3="low ╳" rank4="ruled out ⌀"/>
  <pitch value="E" duration="384" count="3" rank3="low ╳" rank4="ruled out ⌀"/>
  <pitch value="F" duration="384" count="2" rank3="low ╳" rank4="ruled out ⌀"/>
  <pitch value="Bb" duration="256" count="2" rank3="low ╳" rank4="ruled out ⌀"/>
  <pitch value="A" duration="256" count="2" rank3="low ╳" rank4="ruled out ⌀"/>
  <pitch value="C" duration="128" count="1" rank3="low ╳" rank4="ruled out ⌀"/>
```

Figure 59. Staff 6 notation

“Staff=7” | 33-65 Hz | 06:12 in recording and composition

```
<group staff="7" measures="92,93,94">
  <pitch value="G" duration="2560" count="7" rank3="high ●" rank4="high ●"/>
  <pitch value="F#" duration="640" count="4" rank3="medium ◇" rank4="low ╳"/>
  <pitch value="Bb" duration="256" count="2" rank3="low ╳" rank4="ruled out ⌀"/>
  <pitch value="A" duration="128" count="1" rank3="low ╳" rank4="ruled out ⌀"/>
```

Figure 60. Staff 7 notation

4.1.2.2.5. Rhythmic density derived from predominance

There are six categories of rhythmic density. The frame of reference is one second. The number of attacks per second defines the rhythmic category, as expressed below:

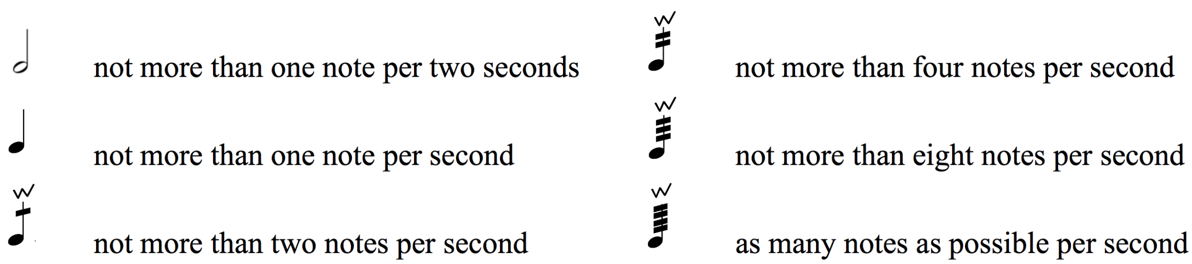


Figure 61. Categories of rhythmic density

As mentioned in the introduction, the rhythmic density categories were derived from the pitch predominance analysis. The “duration” attributes in the pitch-classification listings in the previous sections (4.1.2.2.3., 4.1.2.2.4.) informed the rhythmic density categories. Duration refers to the total time in milliseconds that the pitch is sounding in a segment of time.

The total duration was divided by the total number of seconds of the bars considered in the segment. For example, staff 1 presents C# as the most prominent pitch in measures “24, 25, 26, 27, 28, 29”, with a total duration of 13,184 milliseconds (13.18 seconds) throughout the six measures. The duration per measure is 4 seconds. Therefore, the total duration of the 6-measure segment is 24 seconds. The 24 seconds of the segment divided by the 13.18

seconds in which C# is sounding results in an average rhythmic proportion of 1.8. Therefore, the rhythmic density for bars 24–29 is low, of not more than one note for every 2 seconds. At other points of the piece, the rhythmic density is quite high as the energy of the geyser and therefore its amplitude increases. In this sense, the amplitude contour of the sound model is generally connected to the rhythmic material of the piece.

As noise-derived harmony may result in what Grisey termed as "neutralization of pitch"[10] (Grisey, G. & Fineberg, J., 2000), the rhythmic density categories derived from pitch predominance, as well as spatialization, were implemented not only for the avoidance of monotony, but for perceptual clarity.

4.1.2.2.6. Amplitude analysis

The seven re-synthesized frequency segments processed in SPEAR were imported individually to Logic. Automatically, a waveform display is generated in which the y-axis represents amplitude in dB (decibels) and the x-axis represents time. A screenshot of each waveform display was segmented into seven dynamics regions: ***ppp***, ***pp***, ***p***, ***mp***, ***mf***, ***f***, ***ff***. A drawn contour, shown in various sections of the figures below, was used to track the dynamic evolution of the geyser's frequency regions through time (Figures 62–68). Each of the frequency regions' contours was transcribed to each of the 7 parts of the score.

The dynamic contours of the frequency regions with lowest amplitudes –staff 1 and staff 7, which present the highest and lowest frequencies– were occasionally altered for balance and intelligibility. For example, staff 1, in the figure below, presents a very brief spike at pp, its highest amplitude in the entire 11 minutes of recording. For this reason, a sub-segmentation was made within the ppp range, where the highest peak of is re-interpreted as a mp. In the comparative amplitude analyses below, note the geyser's loudest frequency regions, from 131 to 1047 Hz –staff 3 to staff 5.

---

[10] Grisey was addressing the importance of new techniques that are necessary to avoid monotony.

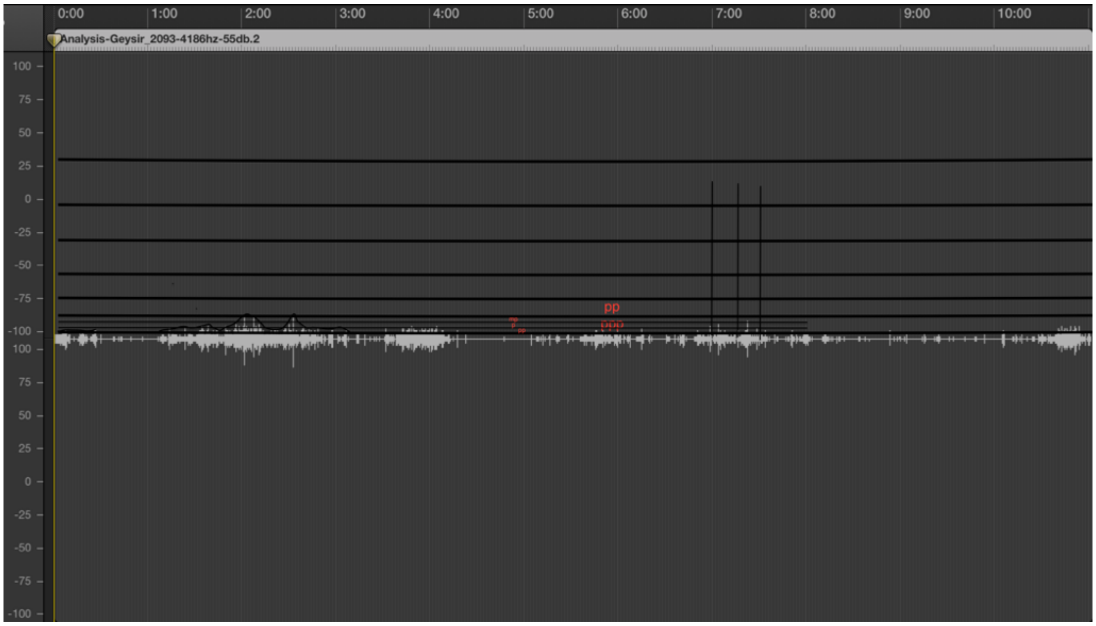


Figure 62. Staff=1 (C7–C8)

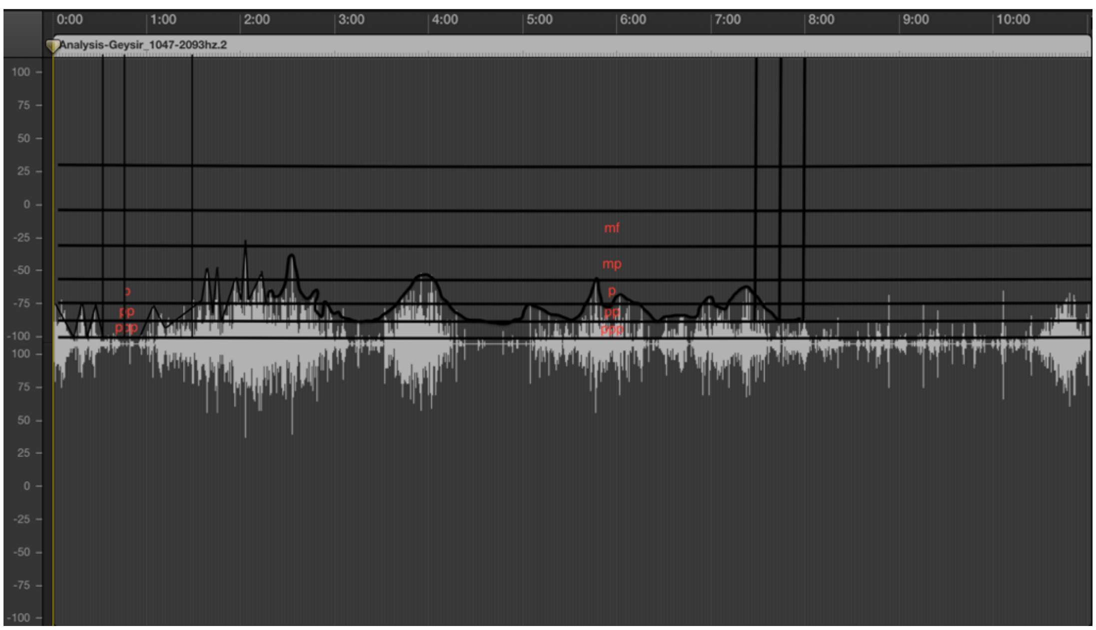


Figure 63. Staff=2 (C6–C7)

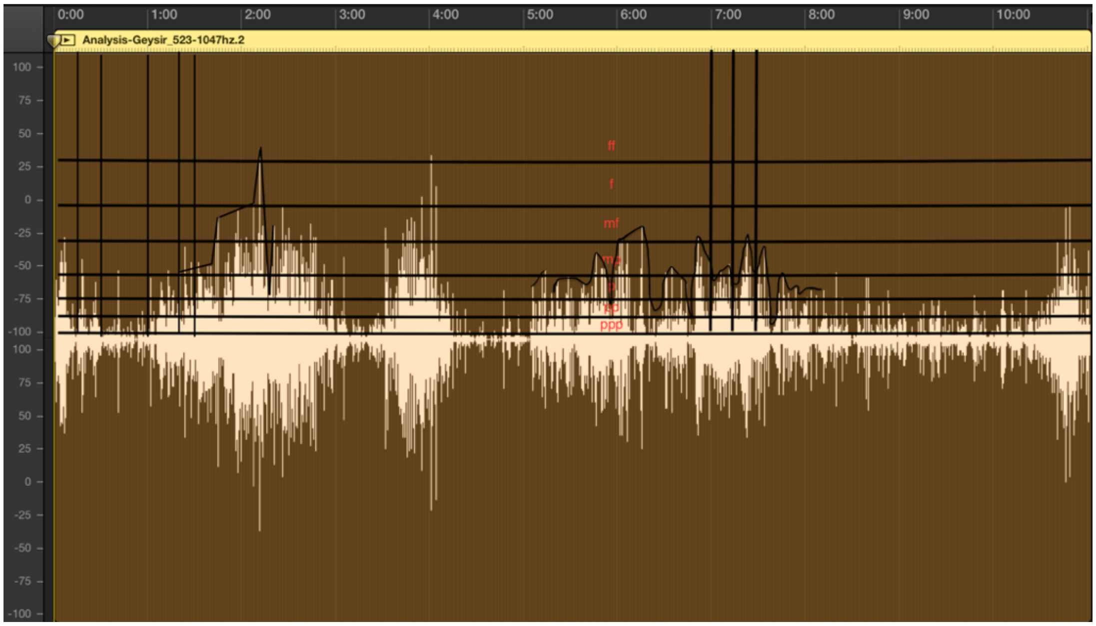


Figure 64. Staff=3 (C5–C6)

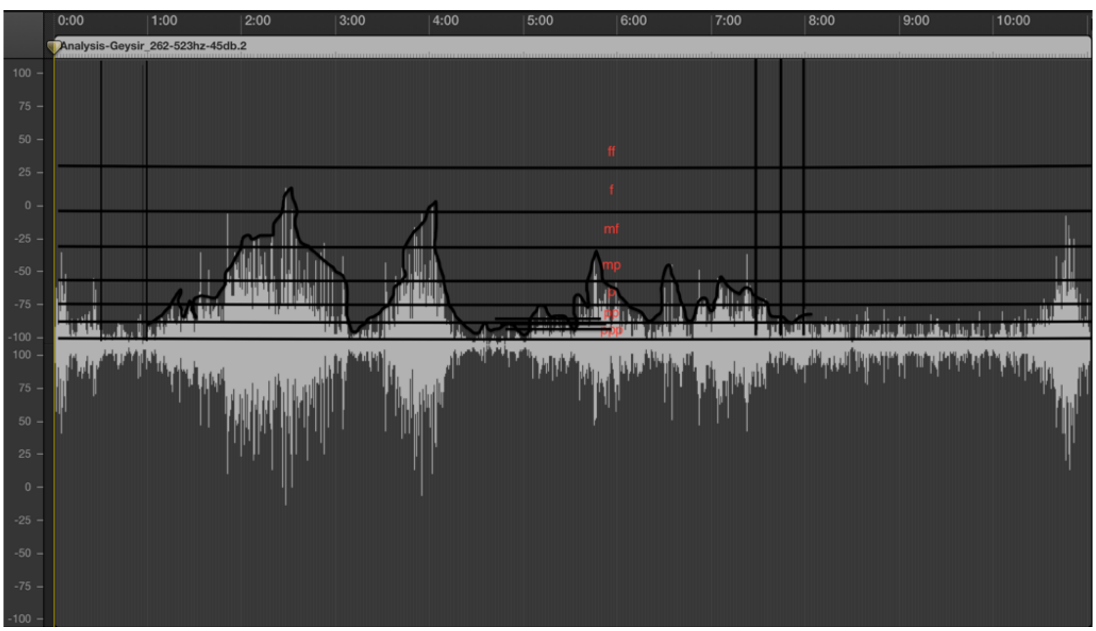


Figure 65. Staff=4 (C4–C5)

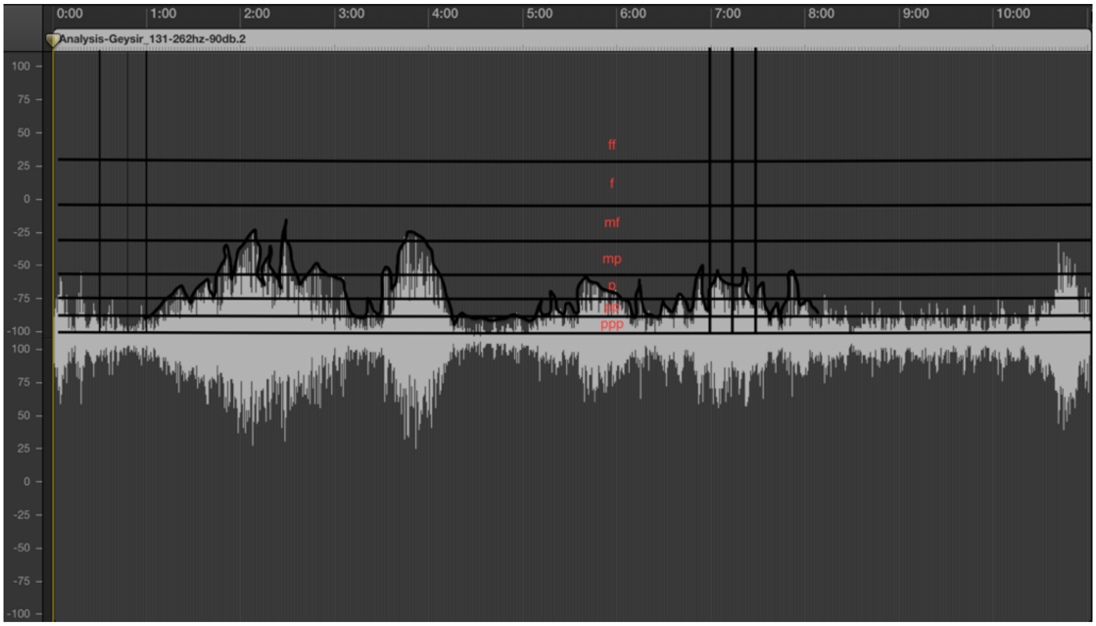


Figure 66. Staff=5 (C3–C4)

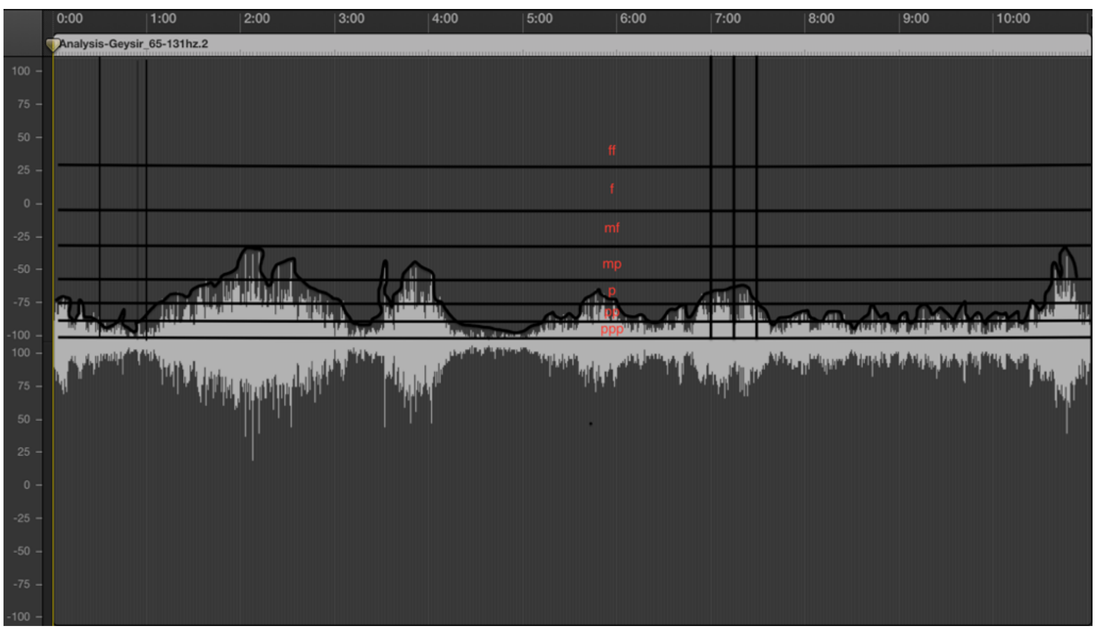


Figure 67. Staff=6 (C2–C3)

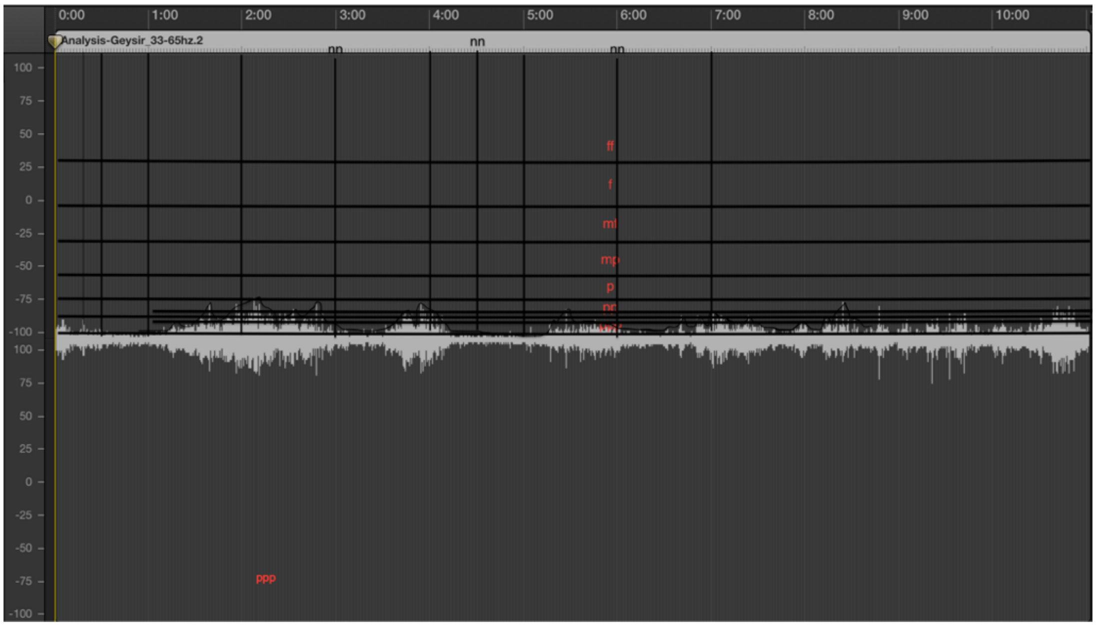


Figure 68. Staff=7 (C1–C2)

In the example below (Figure 69), the full score presents one of the overall peaks in amplitude in the entire 11-minute recording. A close look at the dynamics in each of the instruments will show the correspondences both in the macro- and micro-level of dynamics: while there is a general increase in amplitude from 1:56 to 2:08, there are sudden dips and spikes in the dynamics within the overall increase in the section. The alterations in the dynamics of Staff 1 and 7 are also evident in the example, in which the ***pp*** and ***p***, respectively, are increased to ***mp*** and ***mf*** in order to blend with the dynamics of the rest of the parts.

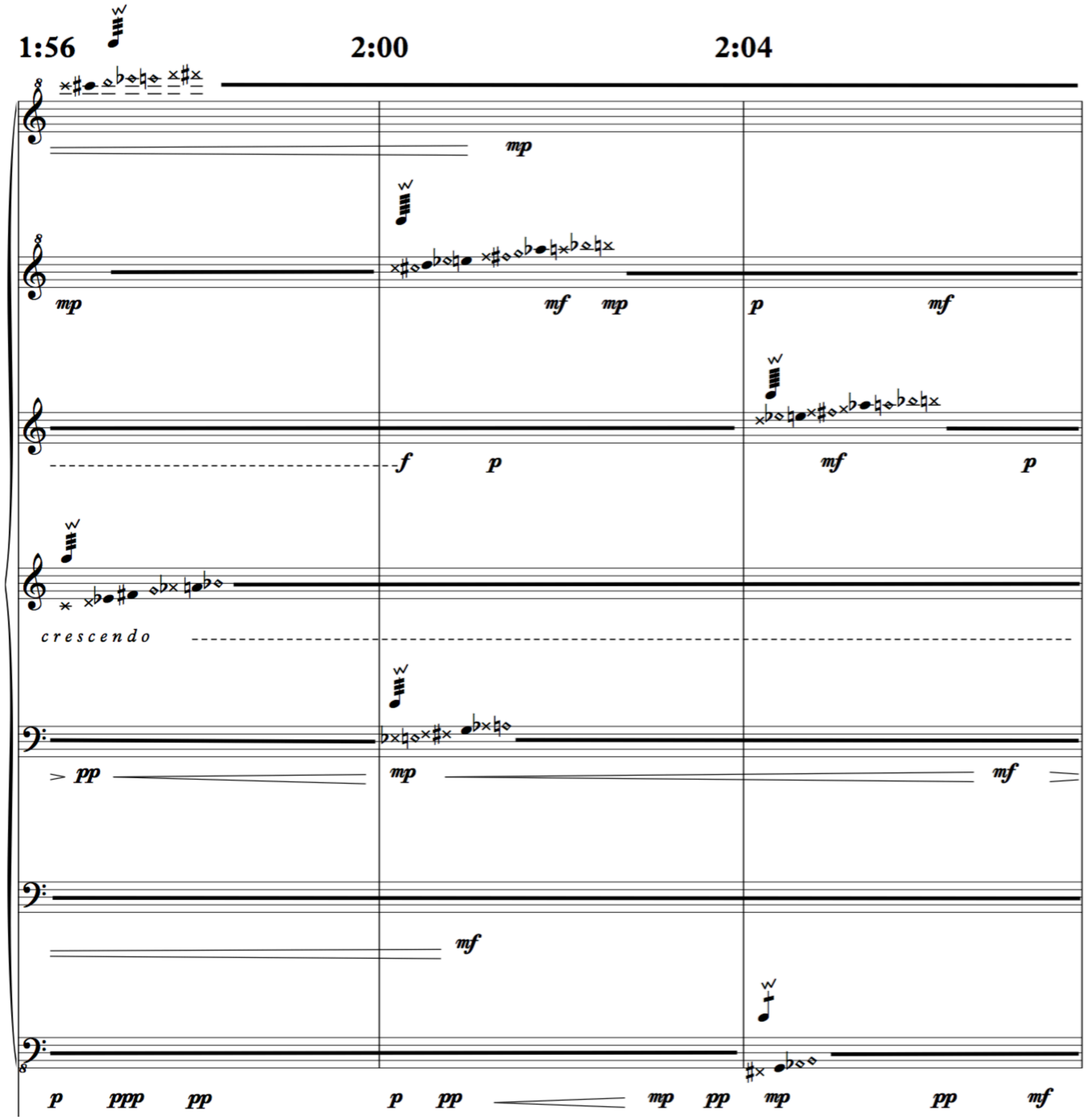


Figure 69. Geyser waveform amplitudes translated to dynamics in full score

4.1.2.3. Notation

As mentioned above, frequency segmentation and pitch-class predominance analyses are motivated by the necessity to reduce the complexity of a broad band noise sound model such as a geyser. This is achieved by means of distilling the predominant pitch content from the noise. In the compositional/analytical section of *Geysir*, I described how, for performance viability, the notation chosen for the music material derived from the predominant pitch sets sought to recreate the aleatoric and chaotic behavior of the geyser's sound model without a high degree of complexity. In *Geysir*, for 7 pianists, each performer follows his/her part with a stopwatch. The pitch-predominance pitch sets (Figures 54–60) are introduced at varying intervals of time (between 12 and 30 seconds). Each pianist plays the pitch sets with varying rhythmic densities depending on the duration of the frequencies registered in the partial tracking analysis. In the pre-compositional stage, high values in frequency duration correlated to high pitch predominance and high rhythmic densities, and vice-versa. In this section, I will explain the procedure to produce motivic content based on pitch and rhythmic density.

In the pitch domain, as explained in the pitch-class predominance analysis key (4.1.2.2.3.), the round note-heads represent high predominance; diamonds represent medium predominance; x represent low predominance. In the rhythm domain, the rhythmic value above the pitch-set determines the density in which the performer will improvise with the pitch set. The symbol placed above the rhythmic value means "irregular" (i.e., a-symmetrically, uneven). The instruction to play irregularly is generalized in the entirety of the score.

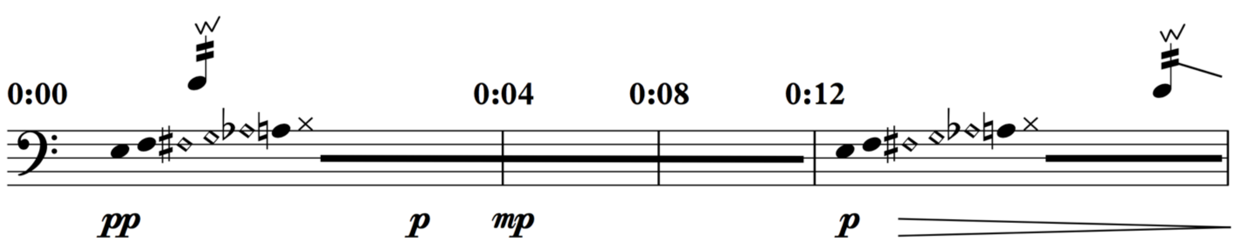


Figure 70. Opening four measures of *Geysir*, piano 5

The performer's choices include 1) the ordering of the sequences of the pitches included in the sets; 2) the durations of the irregular rhythmic values (in the example above, four notes per second, irregularly). The combination of these variables results in a variety of phrases that are generated by the performers, while preserving the essential features of the sound model (i.e., frequency, rhythmic density and amplitude contents).

4.1.2.4. Electronics/Spatialization

4.1.2.4.1. Spatialization

The electronics for the piece consist of seven-channel spatialized fixed media and telematic/live performer amplification. Due to the complexity of the sound model of the geyser and the music generated from it, spreading the streams of audio around the listeners was intended for perceptual clarity. The setup may vary depending on the venue, from a circular distribution of the speakers and all the performers (if the piece is performed in a flat level venue such as a museum), to a half circle distribution of the speakers, two performers on stage and five telematic performers. The latter version is the most viable for traditional concert halls and was the option for the premiere of the piece. The diagram for the setup is the following:

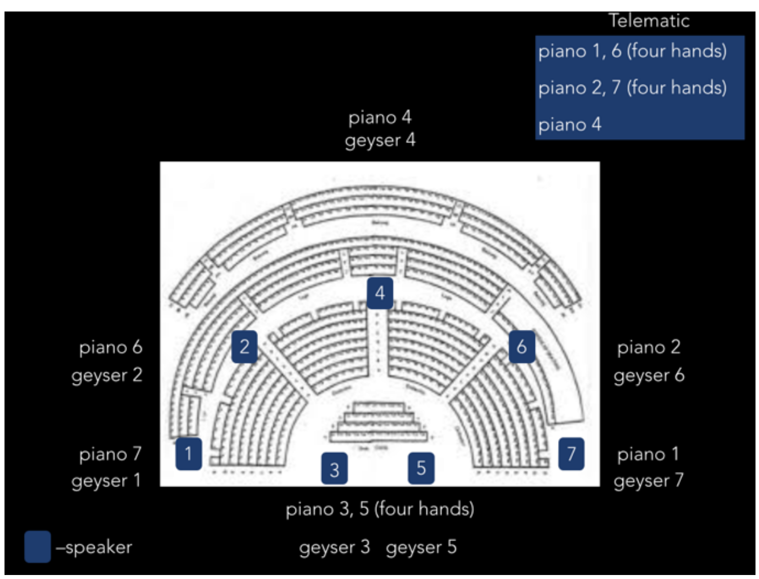


Figure 71. Spatialization diagram

Each speaker projects two sound sources: 1) telematic/live amplified piano; 2) a frequency stratus of the geyser sound model (explained below). There is one piano on stage, to be performed by two pianists playing piano 3 and piano 5, respectively. The other five pianists (piano 1, 2, 4, 6, and 7) performed in piano cubicles situated outside the concert hall, in the University of Virginia's Department of Music. Their sound was sent to the concert hall using XLR cables and their image was broadcast live through a live video chat application projected on a screen on stage.

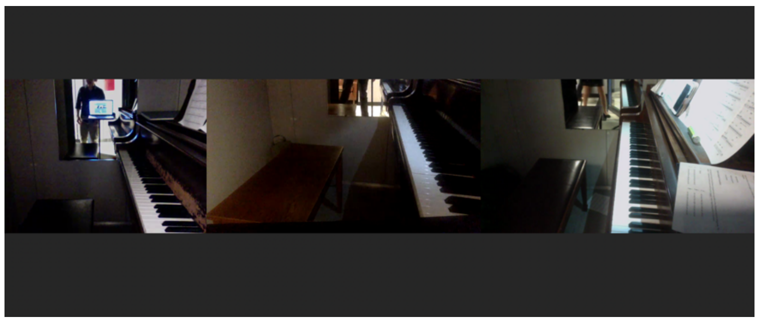


Figure 72. Piano cubicles outside of the performance hall

4.1.2.4.2. Fixed media/instrumental pairing

Each of the seven tracks in the fixed media consists of a specific frequency stratum of the field recording of the geyser from which the piano parts are derived. Each track corresponds to the frequency range of each of the piano parts, based on their octave segmentation (see 4.1.2.3.):

Geyser 1 / Piano 7	Speaker 1
Geyser 2 / Piano 6	Speaker 2
Geyser 3 / Piano 5	Speaker 3
Geyser 4 / Piano 4	Speaker 4
Geyser 5 / Piano 3	Speaker 5
Geyser 6 / Piano 2	Speaker 6
Geyser 7 / Piano 1	Speaker 7

The Spatialization diagram (Figure 71) shows how each speaker projects two different frequency strata. For example, Speaker 1 projects: 1) track 1 of the fixed media (the highest frequency region of the sound model) and 2) piano 7 (the lowest frequency region of the sound model-derive material). This was decided for a balanced frequency distribution throughout the concert hall.

The score presents specific information regarding onsets/offsets of the tracks, as well as dynamics for both the fixed media and the live performers on the mixer.

4.1.2.4.3. Fixed media–live instruments amplitude contour

The seven tracks in the fixed media do are not simultaneously consistently throughout the piece. Each track fades in and out of the mix in order to open the listening field to different frequency regions throughout the piece. When the general amplitude contour of the sound model is low, there is a low number of tracks active. Similarly, although not systematically, a higher number of tracks are active at the points of highest amplitude. For formal direction purposes, the correspondence between amplitude contour of the geyser sound model and number of tracks used in the electronics is scaled differently at the beginning and ending stages of the piece. This is clear in the first and highest amplitude peak (between 1:40 and 2:40), where the maximum number of tracks is three instead of seven, which corresponds to the highest amplitude peak in the second half of the piece.

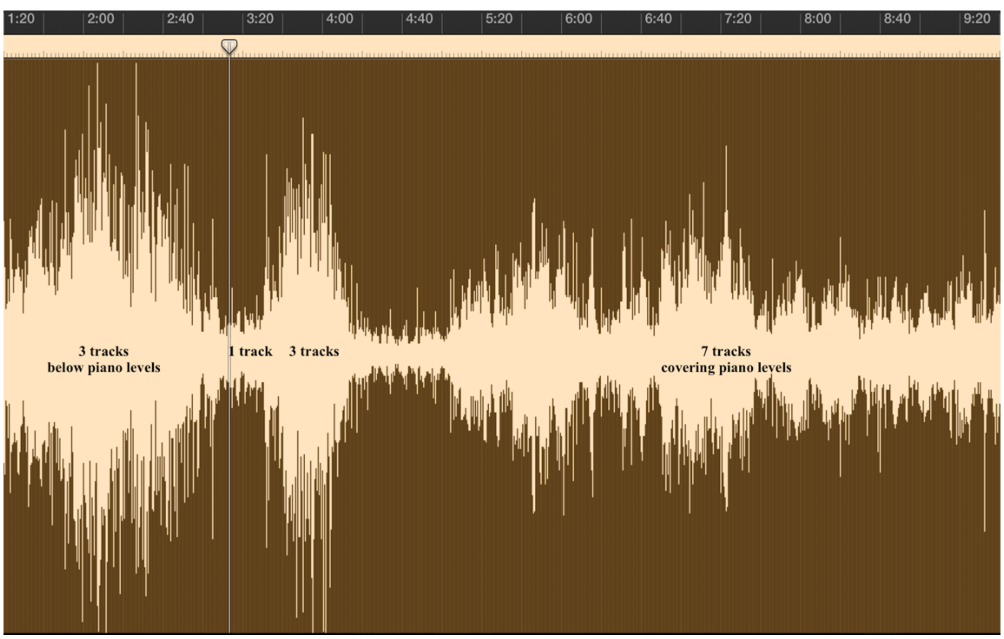


Figure 73. Geysir general amplitude

The overall form of the piece has a general distribution of track density from lower to higher, all tracks being active for the last two minutes of the piece. The simultaneity of all tracks introduces the sound model of the geyser at its full extent in an immersive spatialized environment. As this happens, the dynamics and amplification of the pianos subside, while the levels of the recording of the geyser in the fixed media are increased. In the last minute of the piece, the fixed media is the prevailing sound.

My conceptual and poetic intention behind this formal plan was that the performers, who begin the piece with no electronics, gradually embody the sonic features of the geyser until they have become it. The electronics design is, in this sense, a representation of my philosophy of sound model-based composition –the embodiment of natural principles through the analysis of its sonic features.

4.1.3. Waves Break Aural Shores: FFT size and amplitude threshold variables of a shore sound model as guiding formal principle

This piece is an exploration of the varying ways in which human and technological tools of aural perception can understand a given soundscape. Such perceptual variations depend on several factors: biological, cultural, technological, physical, physiological, psychological, etc. The multiplicity of ways in which reality may be perceived can be experienced when we listen to the same recording several times with varying circumstances (time of day, state of mind, audio device, etc.). In the case of environmental soundscapes—geophony, biophony or anthropophony, pitched or complex—, all of which involve a high degree of intricacy and layers of events, such variability in terms of what is perceived is amplified due to the multiplicity of events that take place simultaneously.

Inspired by Peter Ablinger's *Quadraturen IV ("selbstportrait mit Berlin")* (Ablinger, 1995-98), which reflects on perceptual implications and variances (see chapter 3.2), *Waves Break Aural Shores* was written using various spectral analyses of a field recording of the shore in Puerto Marqués, in the Pacific Coast of Mexico. Each analysis utilizes varying filters, re-synthesis resolutions, registers and loudness thresholds, which result in different musical material, notations and instrumental techniques contrasted by juxtaposition throughout the piece. This compositional idea aims to musically reflect on the notion of different versions of reality (waves) through the multiplicity and variability of perception (aural shores). It is a compositional meditation of Gregory Bateson's idea that "two descriptions are better than one." (Bateson, 1979, p. 67) Underlying the technique and theory behind the piece, my attempt is to translate the breath, pulse, and inner harmony of the breaking waves.

4.1.3.1. Analytical Approach

As explained above, the compositional concept consists of structuring the piece by means of utilizing different versions of analyses and transcriptions extracted from the same sound model (the field recording of breaking waves). This idea determined the procedures that work as foundations of the entire analysis: 1. Filtering of high and low frequency bands; 2. Varying the FFT resolution in higher and lower analysis window sizes. The other two procedures are applied to the resulting analyses from steps 1 and 2 (Filtering and FFT) and consist of implementations of the general analytic procedures for complex soundscapes, explained in 4.1.1., mainly: 3. Frequency segmentation in various registers; 4. Selection of loudest frequencies over a given amplitude threshold.

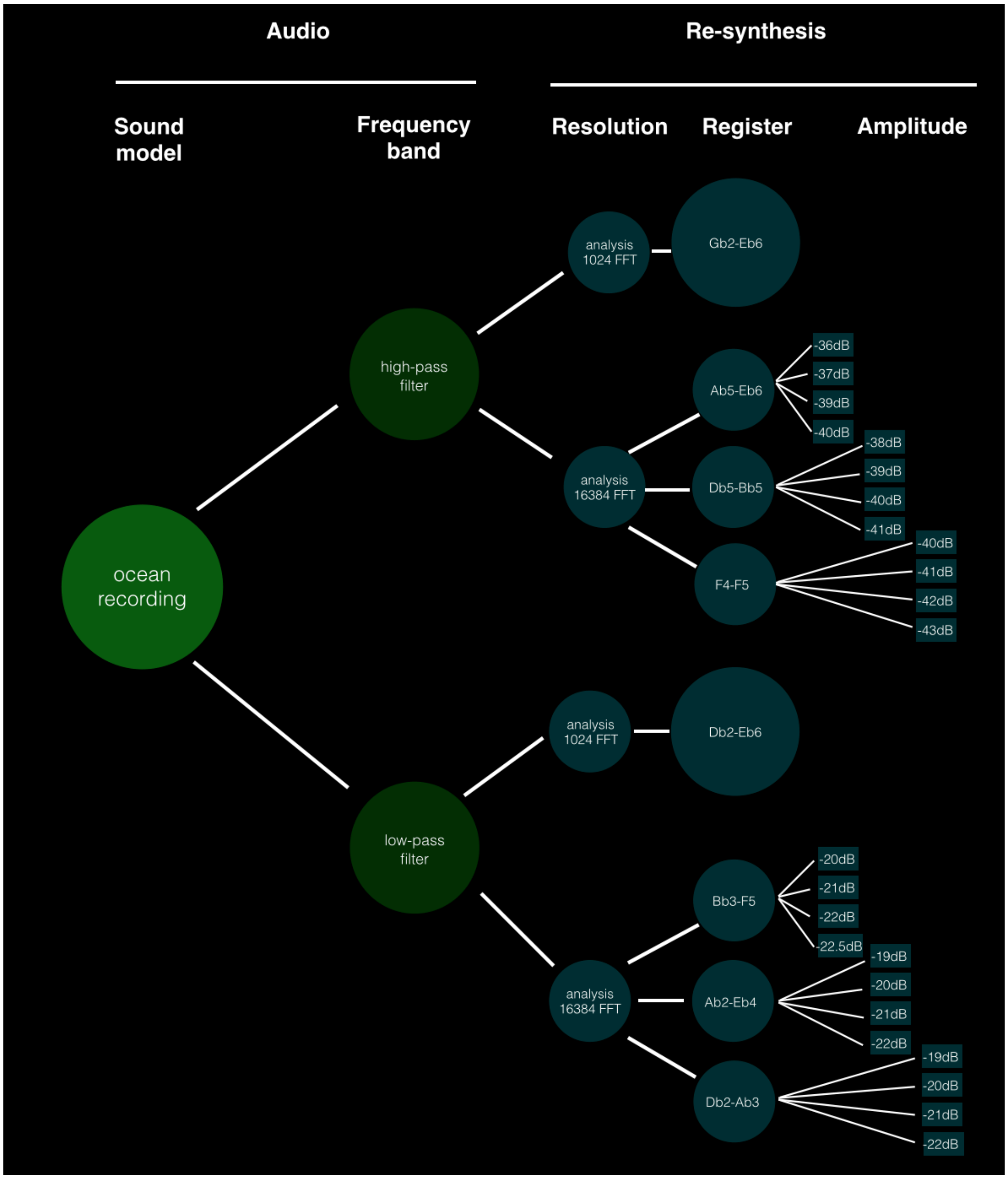


Figure 74. *Waves Break Aural Shores* filtering, FFT, register, and amplitude threshold map

4.1.3.1.1. *Filtering of high and low frequencies*

Partial amplitude differences in the Low-pass and High-pass filtered versions of the original recording derive in complementary harmonic perspectives of the same complex soundscape of the ocean. This is the first degree of analysis in which varying perspectives are considered and generated.

The differences in terms of amplitude contours of the frequency regions are compared in the spectrograms below (Figures 75a and 75b), in which red displays the highest amplitudes and blue displays the lowest amplitudes. The partial with the highest amplitude in the low-pass filtered version peaks at -17 db at 295 Hz. By 2000 Hz, the overall partial amplitude dramatically decreases, peaking at -50 db. In contrast, the partial with the highest amplitude in the high-pass filtered version peaks at -32 db at 1400 Hz. The overall partial amplitude preserves a general energy between -32 db and -39 db up to ~11 KHz, where amplitudes begin to drop.

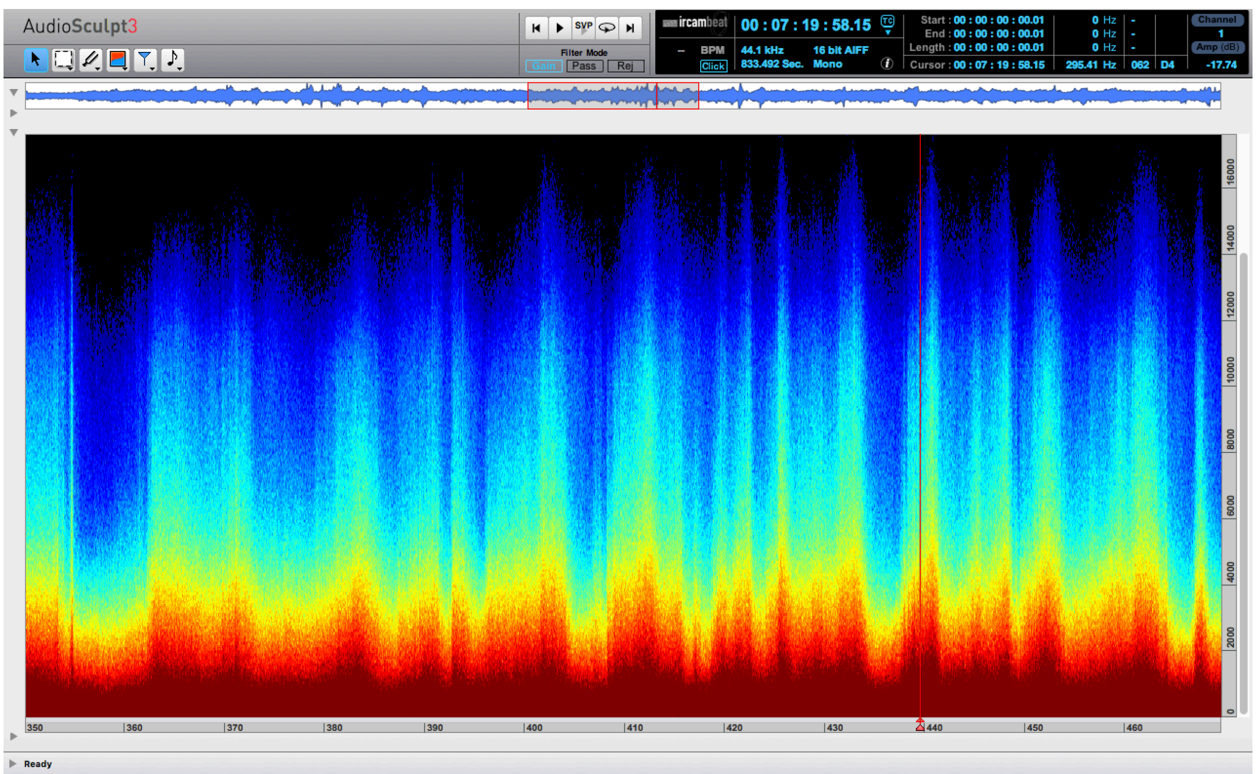


Figure 75a. comparative sonograms
Low-pass filtered version of the ocean breaking waves recording

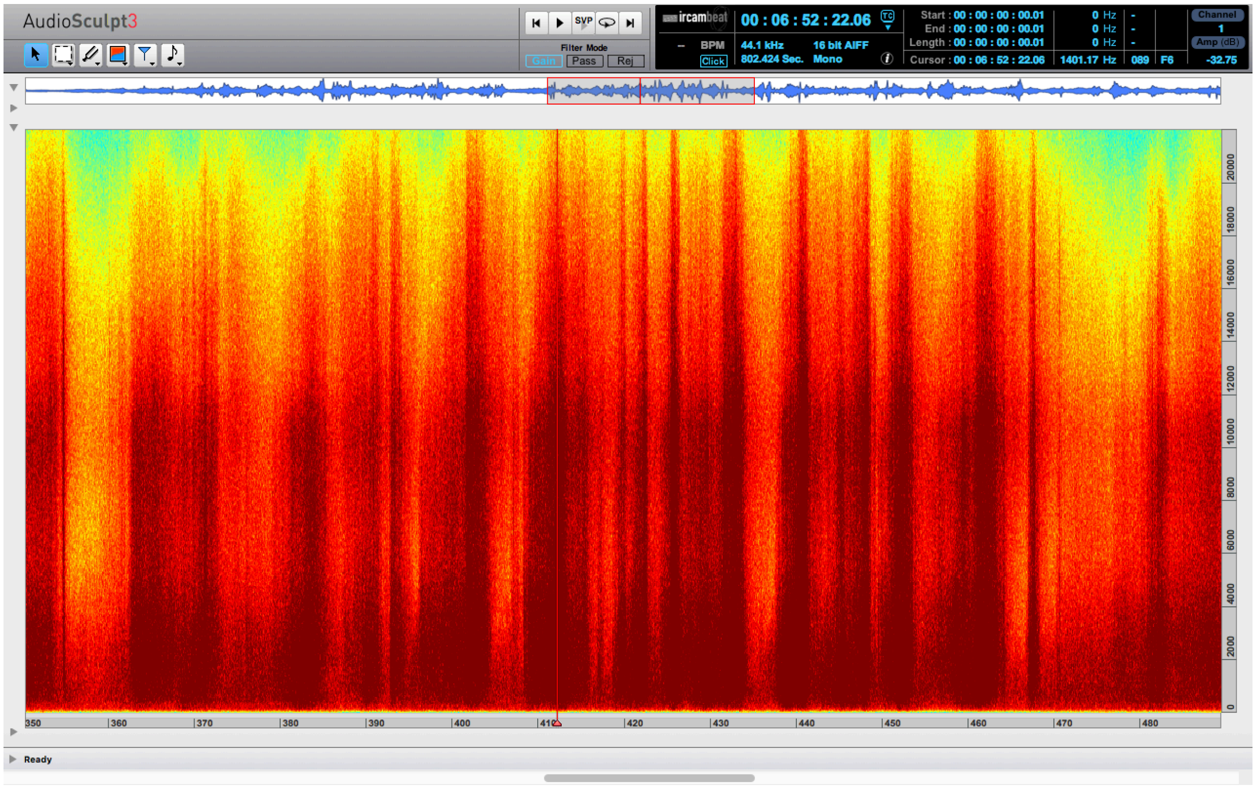


Figure 75b. comparative sonograms
High-pass filtered version of the ocean breaking waves recording

4.1.3.1.2. *FFT size (samples per frame)*

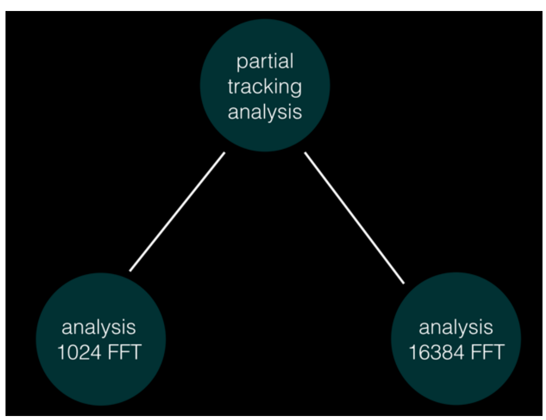


Figure 76. FFT variances

Variations of the number of samples in the FFT constitute the second category of analysis. In a similar vein than that of the High-pass and Low-pass filter versions of the field recording, such variations are driven by the compositional exploration of generating musical material from different FFT analyses of the same recording. As explained above, the purpose is not only an aesthetic exploration, but an effort to disentangle a complex soundscape's inharmonic logic through various and contrasting angles of human/computer perception.

Before explaining the specific FFT procedure for the piece, it is helpful to review the general idea of FFT analysis: "An FFT of a time domain signal takes the samples and gives us a new set of numbers representing the frequencies, amplitudes, and phases of the sine waves that make up the sound that we've analyzed" (Burk et al., n.d., 3.4). For computational efficiency, "rather than looking for the frequency content of the sound at all possible frequencies, we divide up the frequency spectrum into a number of frequency bands and call them bins" (n.d.). The size of the bins is determined by the number of samples in the window or frame. As the window and FFT size increase, so does the frequency resolution.

Resuming the specific FFT procedures for this piece, two contrasting values are applied to the number of samples of the FFT window using SPEAR software: a) 1024 samples per window; b) 16394 samples per window. The former presents bins with smaller duration and longer frequency bandwidth (lower frequency resolution); the latter presents bins with longer duration and shorter frequency band (higher frequency resolution) (Figures 77–78). The ocean field recording is analyzed under these two varying FFT sizes.

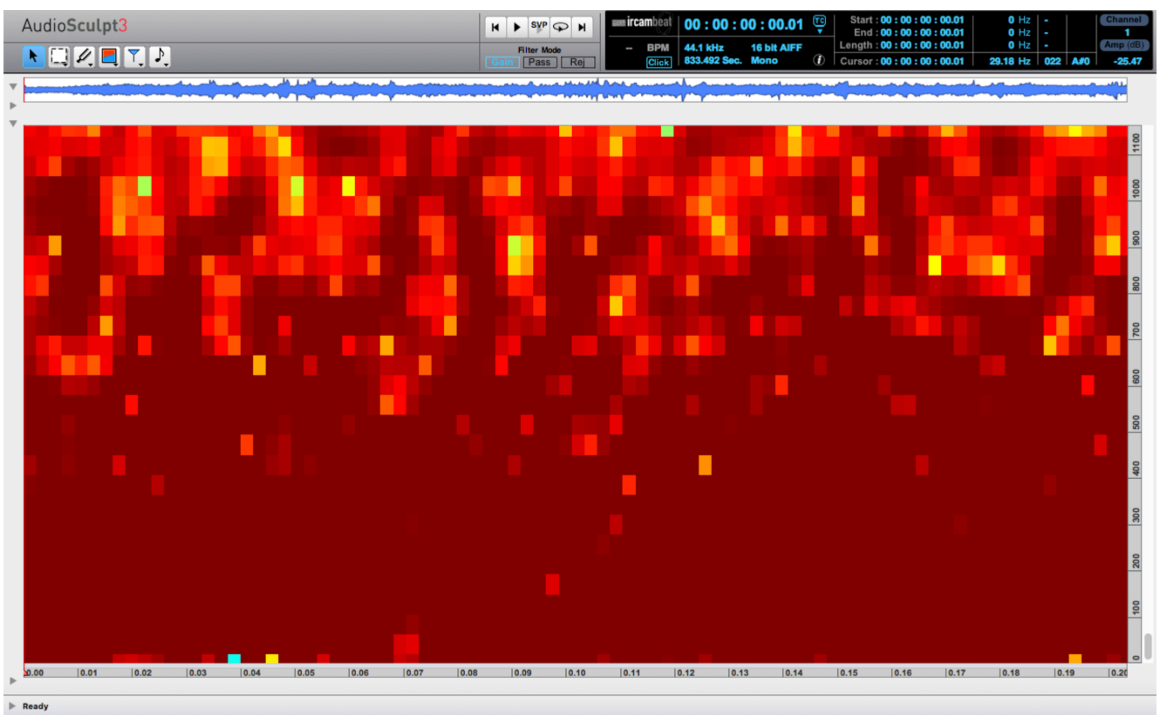


Figure 77. Breaking waves: window size=1024, featuring bins with smaller duration and longer frequency band.

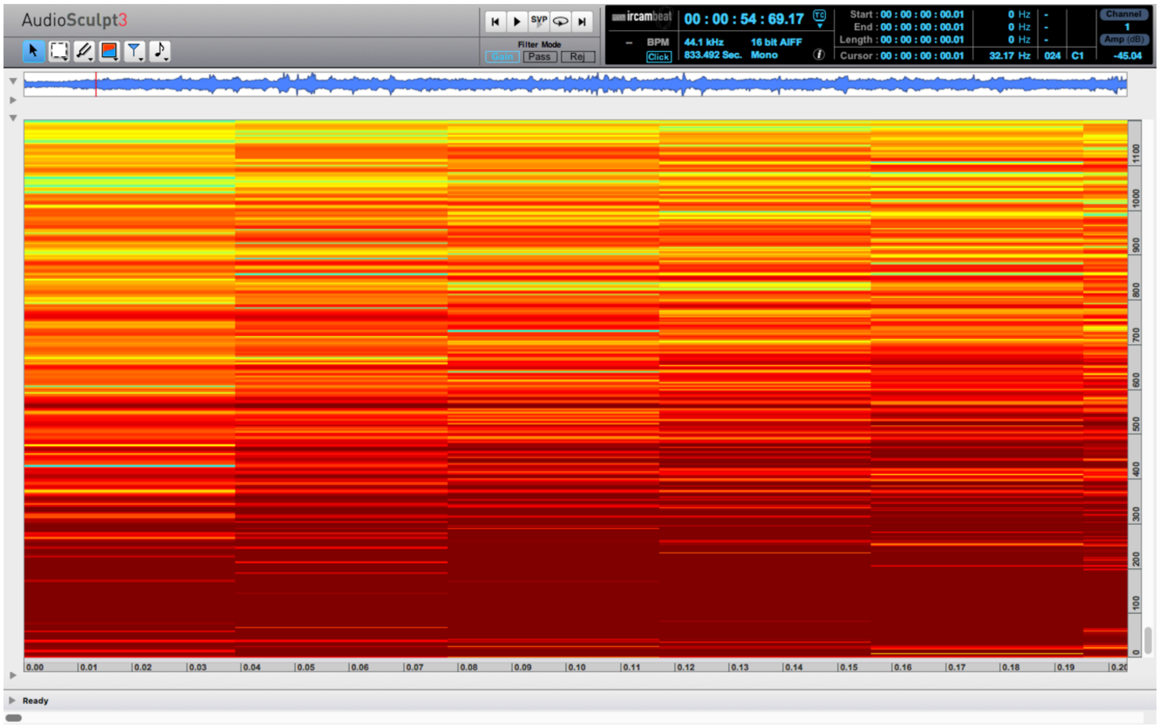


Figure 78. Breaking waves: window size=16394, bins with longer duration and smaller frequency band.

In the case of the former (window size=1024), its smaller duration samples maintain enough resolution to feature the closely adjacent sinusoids. This analysis of the sound model produces material with motives characterized by a high degree of sinusoidal frequency variance (Figure 79 –SPEAR, window size 1024; Figure 80, music notation). On the other hand, the latter (window size=16394) features smaller frequency bands, which show the sinusoidal frequencies in more detail. This higher resolution analysis yields a less volatile sinusoidal frequency behavior as well as rich harmonic material for the piece (Figure 81 –SPEAR, window size 16394; Figure 82, music notation). These musical materials are shown below, first as the re-synthesized sinusoids in SPEAR, and then as the computer assisted transcriptions of the .sdif file into music notation:

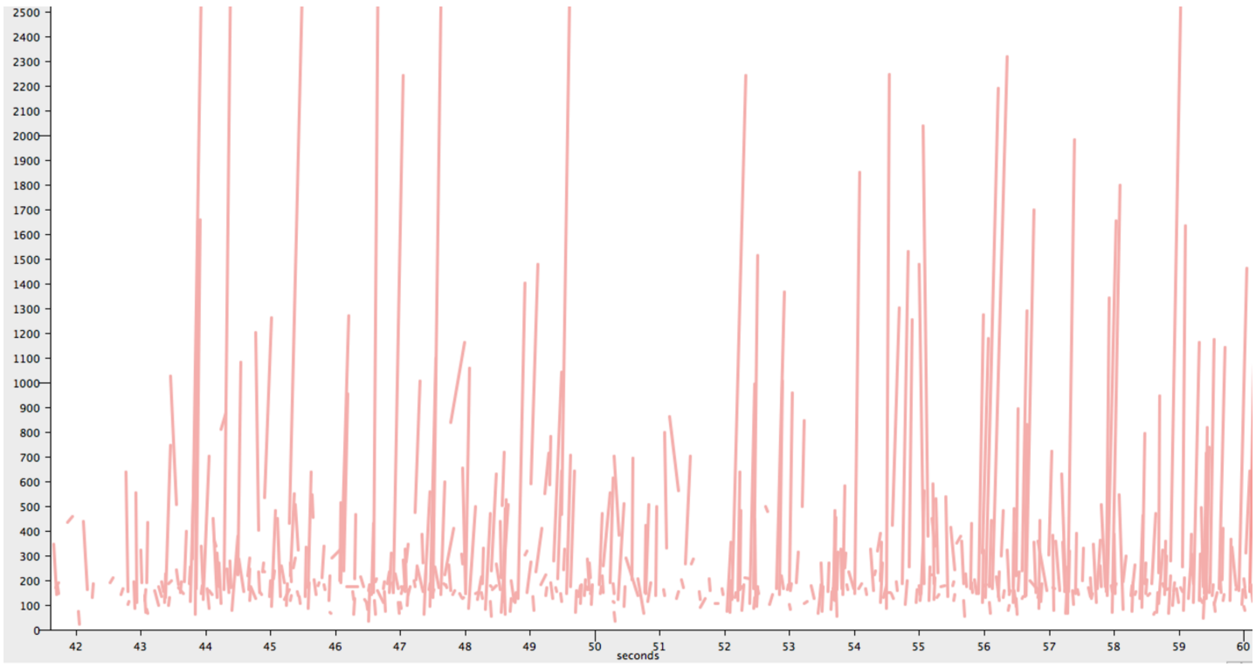


Figure 79. Breaking waves recording, spear re-synthesized .sdif file, FFT size=1024 (loudest partials)

Figure 80. OpenMusic transcription of the re-synthesized .sdif file at FFT size=1024 (loudest partials)

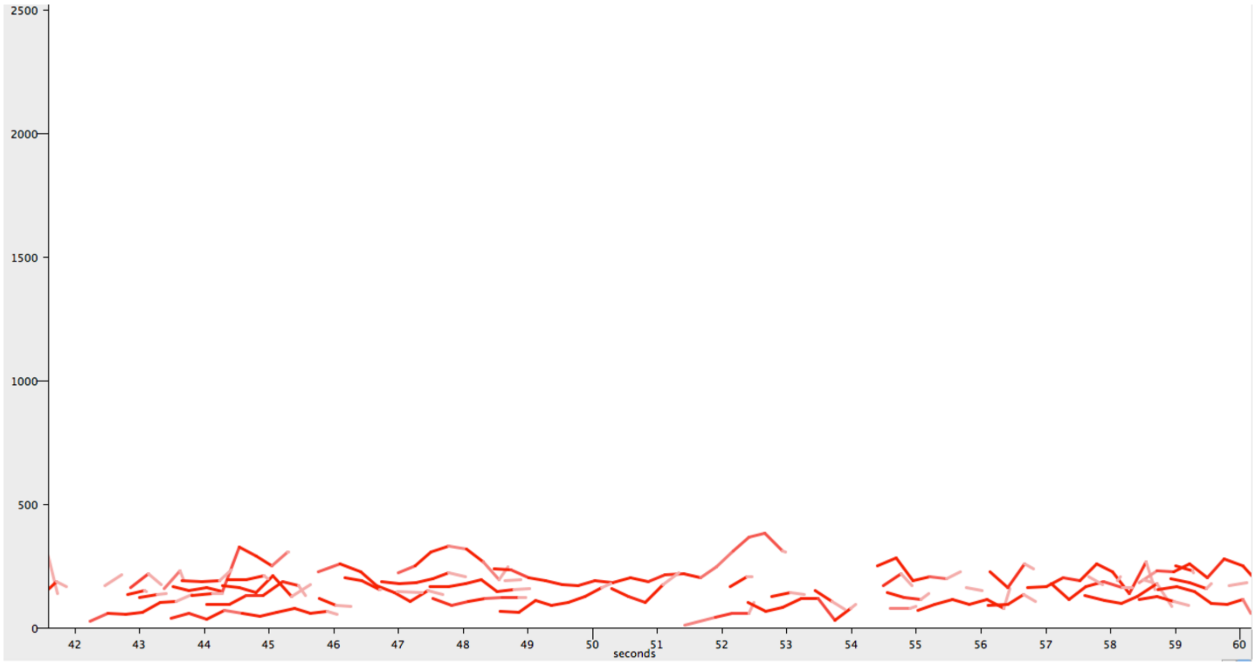


Figure 81. Breaking waves spear re-synthesized .sdif file, FFT size=16384 (loudest partials)

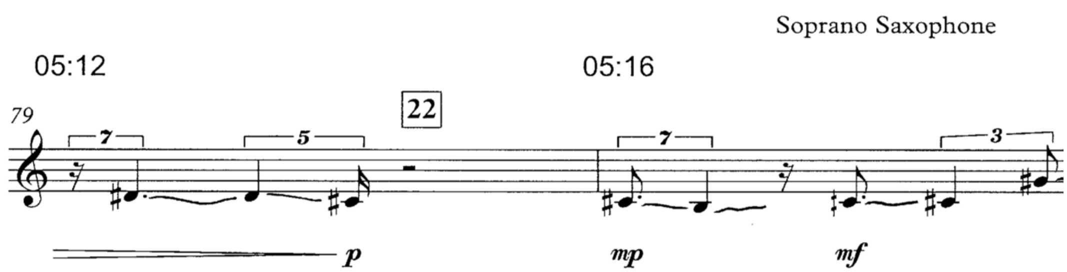


Figure 82. OpenMusic transcription of the re-synthesized .sdif file at FFT size=16384 (loudest partials)

4.1.3.1.3. *Frequency region segmentation and selection of loudest frequencies*

The segmentation of the entire frequency spectrum into different regions is motivated by two necessities derived from transcription and analysis priorities: 1. Polyphonic transcription of the ocean's loudest frequencies per band into the various saxophone registers; 2. A detailed attention to specific time/frequency sections of the ocean recording.

1. The polyphonic nature of a high portion of the piece, in which each of the saxophone quartet parts consists of the loudest partials within a given register, requires segmentation for the OpenMusic (OM) automatic transcription process. The conversion from

a SPEAR generated .sdif re-synthesized files into quantized music notation in OM is simplified if polyphonic voice segmentation is avoided. Since the timbre of the various streams of loudest partials is similar due to the nature of the ocean's sound model, the most feasible process for polyphonic generation is to automatically transcribe one voice at the time in OM. For this reason, frequency region segmentation is essential.

2. The detail that results from focusing on one region at a time is analogous to a microscopic vision which allows us to see specific colors that are otherwise mixed together from a distance. This concept is explored from a compositional perspective, with the intention of presenting the listener various isolated regions of the timbrally complex sound model of the ocean.

Segmentation of frequency regions (see 4.1.1.1.) is applied only to the re-syntheses with a higher resolution FFT size=16384. Frequency region segmentation of lower resolution re-syntheses with FFT size=1024 result in significant alterations in a high number of partials, therefore the procedure was not applied. The frequency regions are based on filter types (low pass and high pass) and the registers of the saxophone quartet. The regions are the following:

High-passed ocean sound model:

- Db5-Bb5 (554-932 Hz)
- F4-F5 (349-699 Hz)
- Bb3-F5 (233-699 Hz)

Low-passed ocean sound model:

- Bb3-A5 (233-880 Hz)
- Ab3-Eb4 (208-311 Hz)
- Db2-Ab3 (69-208 Hz)

4.1.3.1.4. *Varying-amplitude-threshold-analysis-derived melodic material*

In addition to filter types and FFT resolution as criteria for comparisons between varying perspectives of the sound, re-synthesis analyses under varying amplitude thresholds provided different melodic strings of partials at different amplitudes. This procedure allowed flexibility at instrumentation stage, since multiple materials were generated for each frequency region model (i.e., Db2-Ab3, Ab3-Eb4, and so on). At the beginning stage, there was only one voice per register or frequency region, which limited the potential for voice crossings and occasional micropolyphony, important features in *Waves Break Aural Shores*. In this sense, varying amplitude threshold analysis was the solution for deriving melodic material for each frequency region.

In SPEAR, four different ocean re-syntheses were performed for each frequency region sound model under varying amplitude thresholds: -19dB, -20dB, -21dB, and -22 dB, respectively (see 4.1.1.2.2.). The resulting re-synthesized files are four layers of streams of partials from a lower amplitude background of -22 dB to a higher amplitude foreground of -19 dB. Each of these .sdif files are converted into music notation in OpenMusic. As an example, the frequency region of Register 1 (69–208 Hz / Db2–Ab3) generated the following pre-composition material for the first 20 seconds of the piece (Figures 83–86):

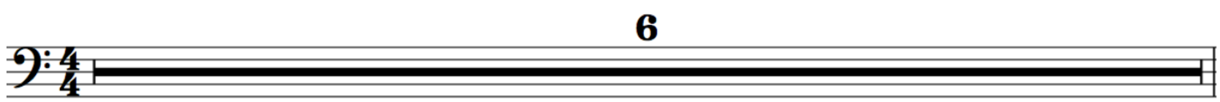


Figure 83. Register 1 (69–208 Hz / Db2–Ab3), -19 dB

Figure 84. Register 1 (69–208 Hz / Db2–Ab3), -20 dB

Figure 85. Register 1 (69–208 Hz / Db2–Ab3), -21 dB

Figure 86. Register 1 (69–208 Hz / Db2–Ab3), -22 dB

In the figure below (Figure 87), a full score with the entire re-synthesis-derived melodic material per frequency region is shown. Each box on the left includes the four instruments that constitute the saxophone quartet (baritone, tenor, alto and soprano saxophones), with their corresponding melodic material derived from the varying re-synthesis amplitude thresholds. The upper end of the score includes the frequency regions derived from the high-pass filtered sound-model of the ocean, while the lower end includes those derived from the low-pass filtered model. The blue boxes show the material derived from analyses with an FFT size of 1024 samples per window; the purple boxes show material derived from analyses with an FFT size of 16384 samples per window. The score worked as a motivic and register palette from which materials were selected for the various sections of the piece.

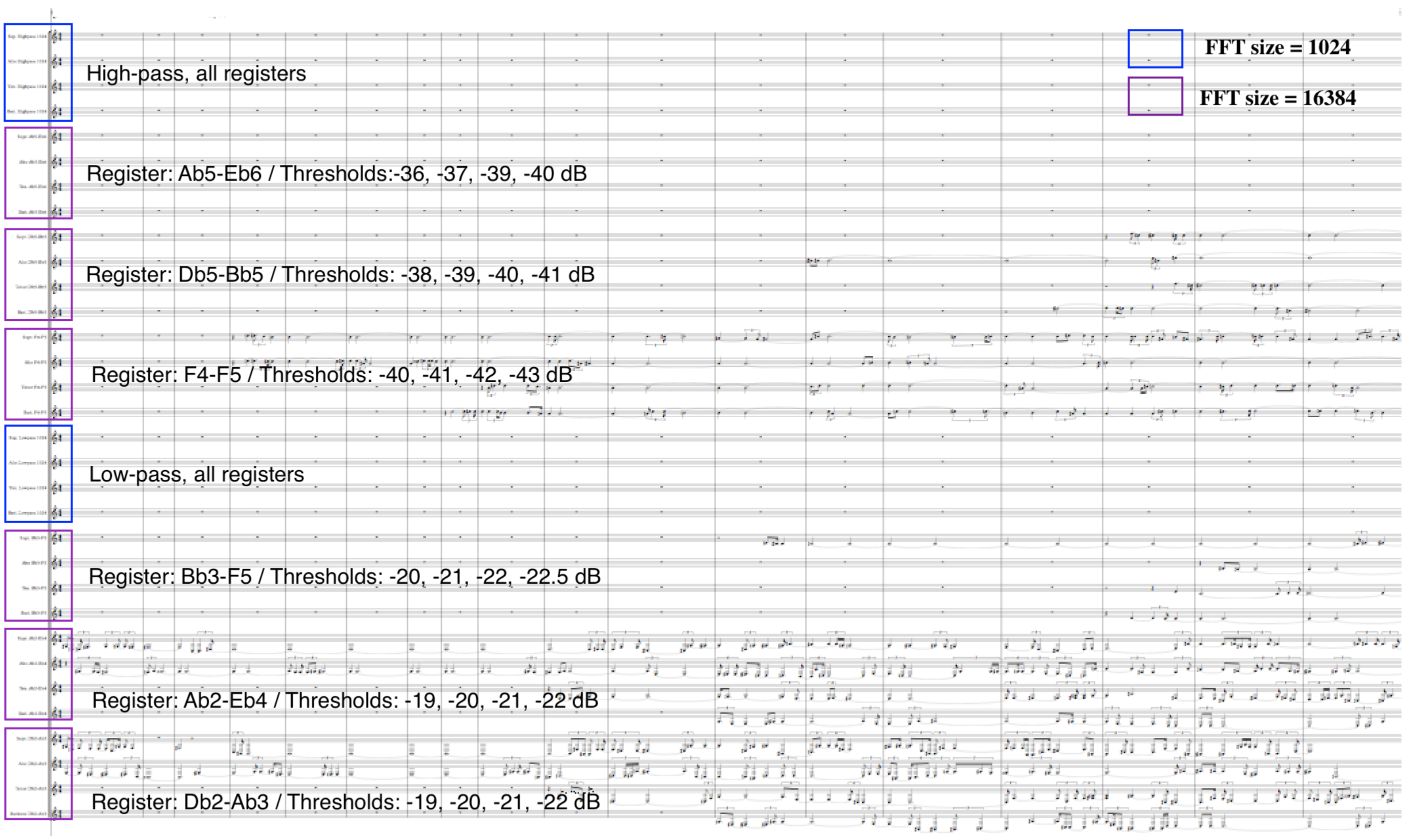


Figure 87. *Waves Break Aural Shores* full palette of re-synthesis-derived melodic material per frequency region

4.1.3.1.5. *Peak amplitude categories*

As explained in 4.1.1.2.1., form and material-type decisions were based on peak amplitude measurements of each the forty-eight wave cycles in the ~10-minute field recording of the ocean. In AudioSculpt, markers were placed by hand to label each wave cycle. The markers were exported as a MIDI file in order to instantly locate each wave cycle in the amplitude analysis (Figure 88). The MIDI file was imported to a DAW (Logic X), where peak amplitudes of each wave break were manually-aurally determined using the peak level display (Figures 89a–b). The peak amplitudes derived in a list and a graph that was the basis for the formal plan of the piece. Similar to the audio waveform display in the DAW, the graph displays the varying amplitude regions of the piece in time, adding the precise values of each peak amplitude (Figure 90).

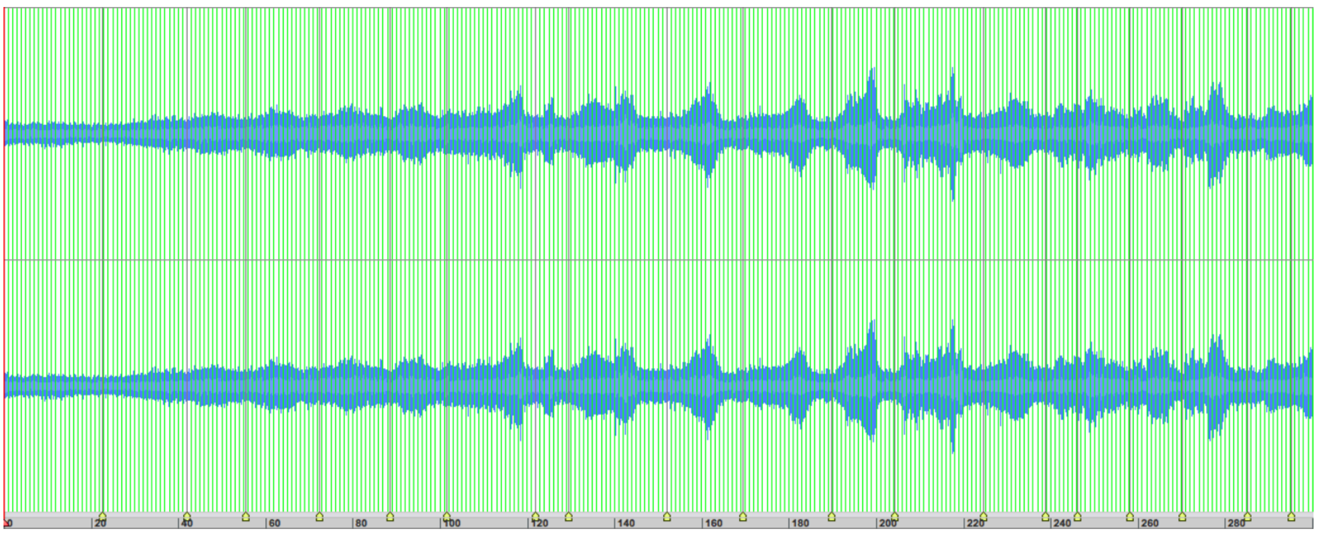
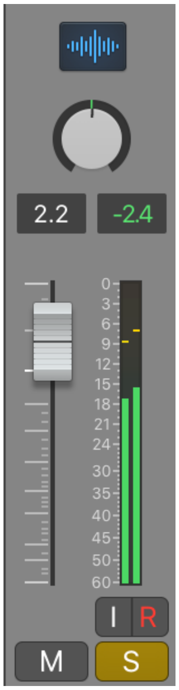


Figure 88. Ocean recording audio waveform with midi markers dividing each wave cycle, 00:00-05:00 (AudioSculpt)



Section/Wave	Time code	Peak amplitude	Section/Wave	Time code	Peak amplitude
1	00:00.0	-11	25	06:15.1	-3.3
2	00:22.6	-7.9	26	06:27.0	-6.1
3	00:42.0	-7.1	27	06:36.3	-5
4	00:55.3	-5.4	28	06:46.2	-3
5	01:12.2	-5.9	29	06:58.2	-4.1
6	01:28.5	-6.2	30	07:03.8	-2.7
7	01:41.5	-2.9	31	07:07.3	-3.1
8	02:01.7	-5.5	32	07:15.6	-2.5
9	02:09.5	-1.9	33	07:22.9	-3.6
10	02:31.9	-3.1	34	07:30.0	-2.2
11	02:49.3	-4.9	35	07:35.8	-4.8
12	03:09.8	-0.5	36	07:45.3	-5.9
13	03:24.2	0	37	07:49.8	-4.9
14	03:44.6	-2.8	38	08:05.8	-3.6
15	03:58.7	-6.8	39	08:16.8	0
16	04:06.0	-1.6	40	08:24.2	-3.1
17	04:18.0	-4.8	41	08:41.8	-5.9
18	04:30.1	-1.2	42	08:51.0	-5.5
19	04:44.9	-7	43	09:03.0	-5
20	04:55.0	-4.4	44	09:14.6	-6.7
21	05:03.0	-4.1	45	09:23.6	-1.3
22	05:14.0	-6.3	46	09:33.4	-6.1
23	05:23.8	-4.9	47	09:40.0	-5.5
24	05:57.1	-5.4	48	09:51.3	-5

Figure 89a

Figure 89b

Figure 89a–b.
a) manual-aural amplitude transcription using the peak level display in Logic X;
b) wave break peak amplitudes (in decibels)

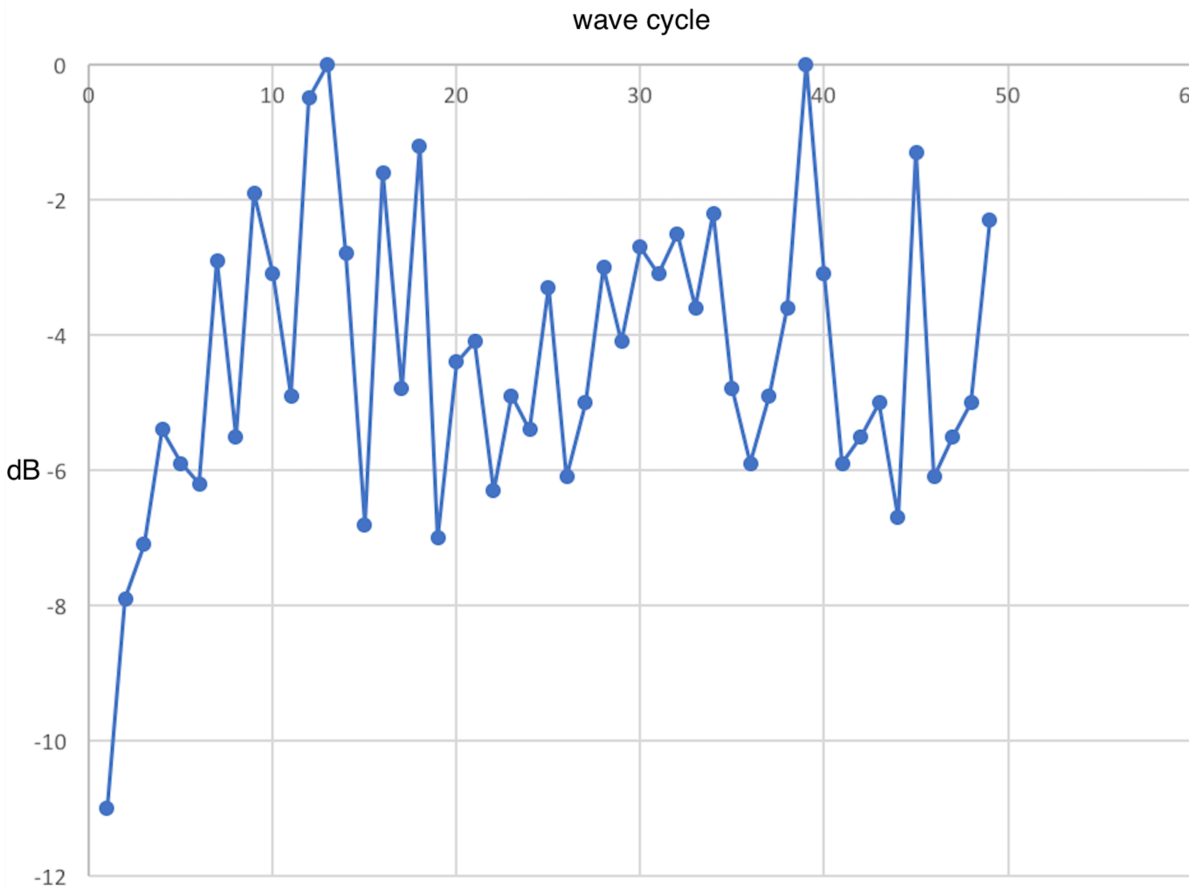


Figure 90. Peak amplitude display of 48 wave breaks in the ocean recording

The amplitude data was then organized in eight categories of intensity, 1 being the lowest amplitude and 8 being the highest. Each of these categories was associated to specific dynamic regions (***p*** to ***fff***) as well as to various musical materials derived from the re-synthesis analyses discussed above and from other compositional approaches (Figure 91). Through this methodology, the varying cycles of intensities –amplitudes– in the ocean's breaking waves is translated to the formal procedures of the piece.

The pairing of the re-synthesized analyses with an FFT of 1024 samples per window with the highest amplitude categories (i.e., Category 8–FFT=1024, ***fff***) was the starting point for the peak-analysis-derived sections of the piece. On the basis of building the piece from the climactic points, the high degree of sinusoidal instability that characterizes the material derived from the 1024 samples/window re-syntheses, was ideal for the intensity of the highest intensity breaking waves. On the other end of the spectrum, the material derived from the

16384 samples per window re-syntheses, with more sinusoidal and therefore harmonic stability, was used both for the low and high amplitude categories (i.e., 2, 3, and 6).

In addition to the re-synthesis derived material, other timbral explorations were composed and mapped to the categories: a) Aeolic sounds produced by blowing directly to the body of the saxophones (without the mouthpiece), which produce a sound quite similar to the sound of a distant shore; b) Multiphonic glissandi, produced by each performer playing a staff of saxophone notations while singing another staff of vocal material (derived from the varying-amplitude-threshold-analyses); and c) Multiphonics, which are determined based on lists of average frequencies of the sections of the ocean recording in which the multi-phonics are used (explained below with more detail). These materials will be explained in 4.1.3.2.

Amplitude category	Instrumentation (sax quartet)	Dynamics
1	aeolic sounds	*p (ff)*
2	monophonic glissandi	*mf*
3	polyphonic glissandi	*mf*
4	1024 resolution	*mf*
5	multiphonics	*f*
6	polyphonic glissandi	*ff*
7	multiphonics	*fff*
8	1024 resolution	*fff*

Figure 91. *Waves Break Aural Shores,* wave breaks peak amplitude categories matched with saxophone material

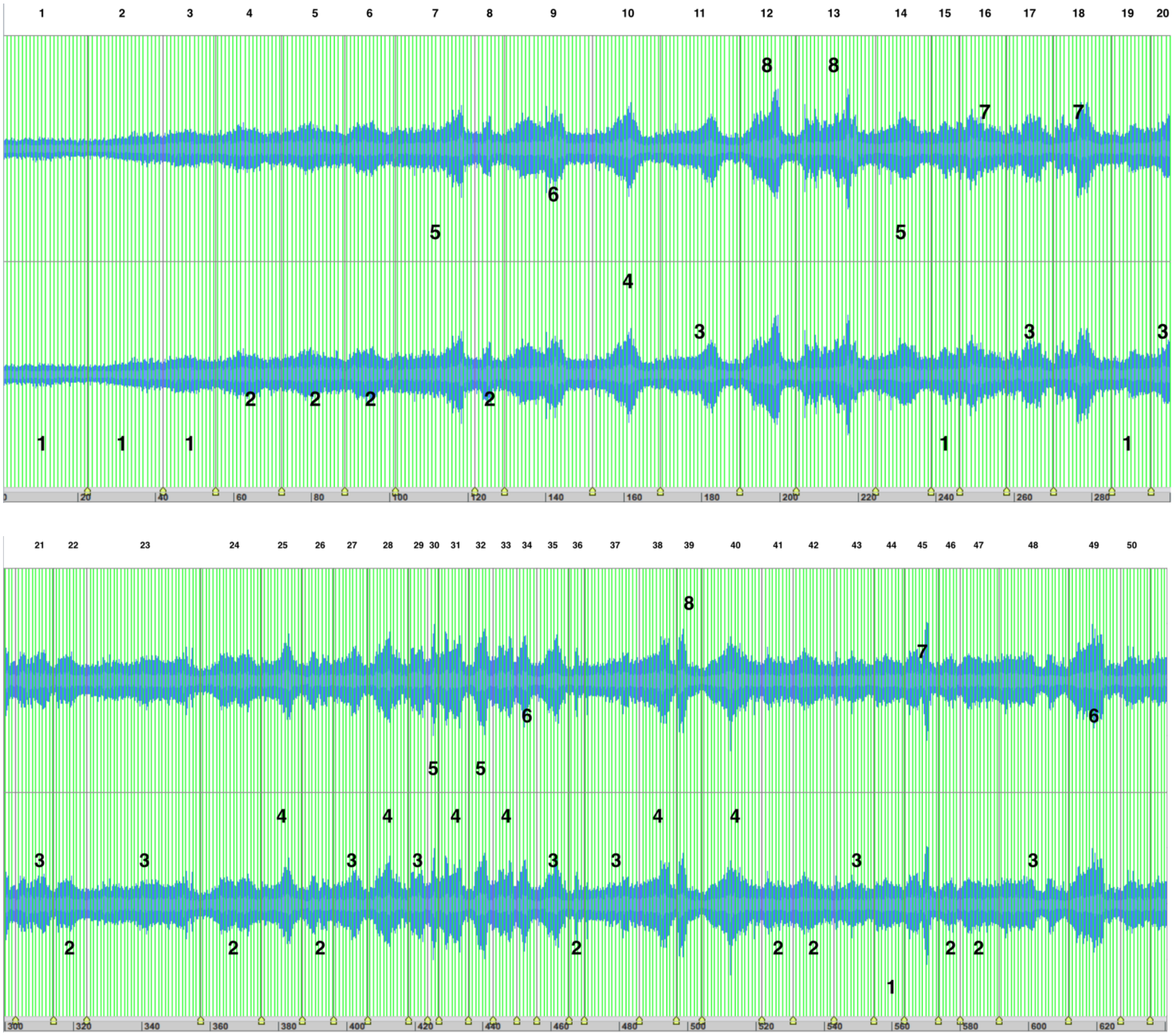


Figure 92. *Waves Break Aural Shores*, distribution of peak amplitude categories over the ocean recording.

4.1.3.2. Sound model-derived material

After the computer-assisted transcription of the .sdif re-syntheses into music notation in OpenMusic, some adjustments were made in order to create a higher degree of connection to the nature of the sound model (the ocean). This may be understood as a return to the sound world of nature after having synthesized it for analytic and transcription purposes. This natural sound world implies the performer's intuition. The notations used are flexible enough to produce musical fluidity, while at the same time precise enough to maintain the acoustic data from the FFT analyses as a foundation for harmony, dynamics and timbre.

Besides the music materials derived from the analyses, a material that was developed more intuitively—the Aeolic sound sections—will be discussed.

4.1.3.2.1. *Monophonic glissandi derived from FFT=16384*

This is the most predominant material in the overall form of the piece, as seen above in Figure 91—*Waves Break Aural Shores, distribution of peak amplitude categories over the ocean recording*. It is derived from the process described in the Analytical Approach (4.1.3.1.). Specifically, this material is derived from an FFT analysis set to 16384 samples per window which, as mentioned in 4.1.3.1.2., yields a stable frequency behavior as well as rich harmonic material for the piece when presented in a 4-voice polyphony.

The sound model of this material is the constantly pitch-bending sinusoids resulting from the FFT re-syntheses. In order to preserve the sinusoidal behavior in the sound model, glissandi were added to the determinate notations derived from the OpenMusic transcriptions. The instructions for the performers are to perform the glissandi *ad. libitum*, according to their technique and characteristics of their instruments. The combination of the indeterminacy in the glissandi and the determinacy in the rhythms and pitches results in a balance between the grid-like analytically re-synthesized domain and the organized randomness introduced to nature by the agency of its individuals.

Figure 93. *Waves Break Aural Shores*, monophonic glissandi

4.1.3.2.2. *Polyphonic glissandi derived from FFT=16384*

The origins and nature of this material are exactly the same as those of the monophonic glissandi. For this material, however, the performers produce two different sounds simultaneously: saxophone sounds and sung sounds. Each performer reads from two different staves: 1. An upper staff with the normal transposed notation for the saxophones; 2. A lower staff with notation “in C”, to be sung by the performer while emitting air through the saxophone mouthpiece. The registers were decided based on those of the performers of the New Thread Quartet.

The idea for these polyphonic glissandi originated from a textural necessity, as an intent to expand the number of voices of the saxophone quartet due to the nature of the ocean sound model.

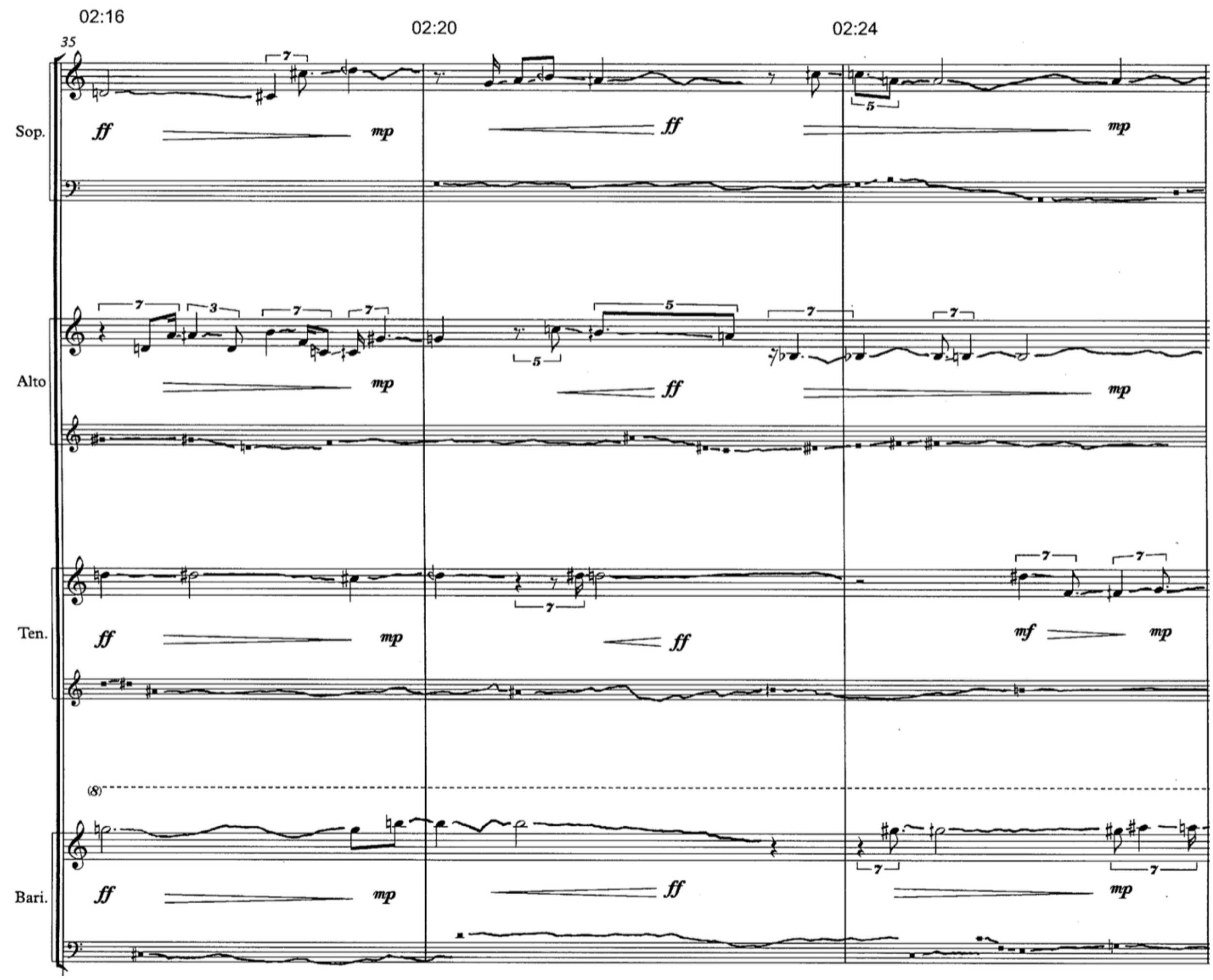


Figure 94. *Waves Break Aural Shores*, polyphonic glissandi

4.1.3.2.3. *Pointillistic saturation derived from FFT=1024*

The high degree of sinusoidal frequency and register variance, characteristic of the analyses set to FFT=1024 frames per window, yielded material with high rhythmic complexity. Proportional notation was derived from the OpenMusic .sdif transcriptions' determinate notation. This was motivated by an interest in feasibility and fluidity—instead of focused rationalism—of performance. In the example below, the original determinate notation from OpenMusic is transformed into proportional notation.

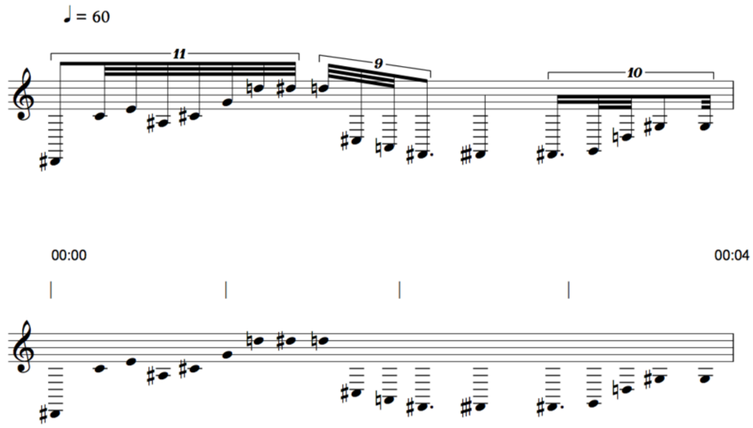


Figure 95.
Above: OpenMusic-derived determinate notation
Below: Proportional notation derived from the latter

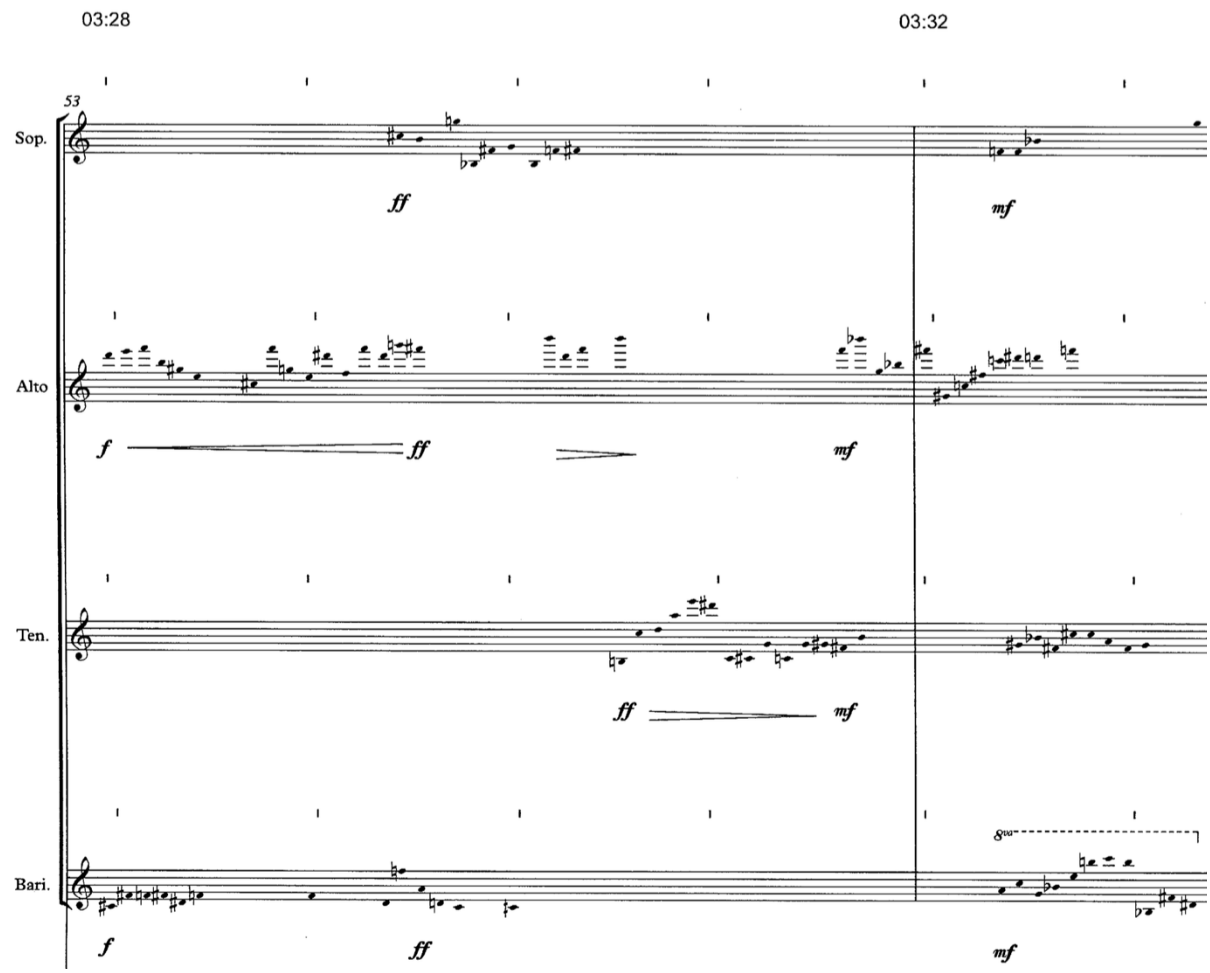


Figure 96. *Waves Break Aural Shores*, section 13, pointillistic section

4.1.3.2.4. *Multiphonics derived from loudest average frequencies*

The multiphonics sections in *Waves Break Aural Shores* aim to achieve the highest possible harmonic density with the saxophone quartet and the closest possible sonority to the sound model of the ocean. In SPEAR, average frequency snapshots of the ocean's breaking waves recording were taken at peak amplitude events from categories 5 and 7 (for reference of the peak amplitude categories, see 4.1.3.1.5.). The average frequency snapshots included the loudest sinusoids in a given point in time, between -2.9 and 1.2 dB. Once the snapshots were taken, lists of chords were generated. Multiphonics were chosen for each instrument according to their pitch content similarities to the average frequency chords derived from the ocean recording.[11] The following chords are derived from the ocean's average frequency snapshots at peak amplitude categories 5 and 7:

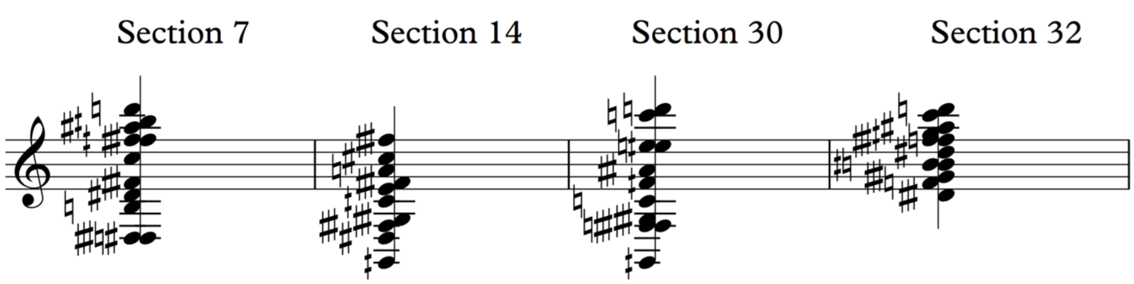


Figure 97. Peak amplitude category 5 list of chords

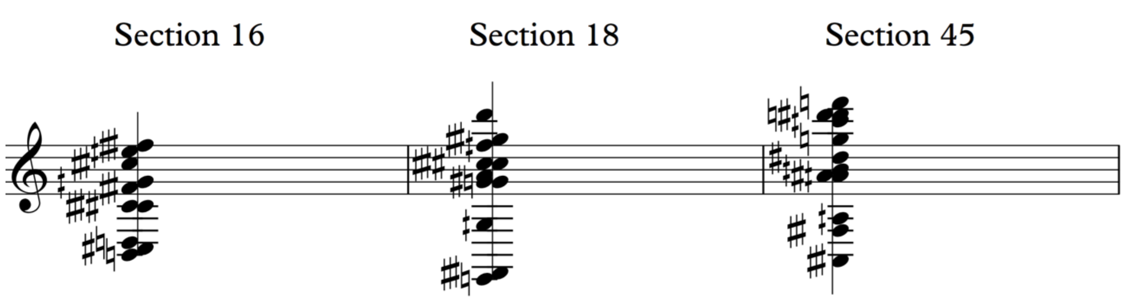


Figure 98. Peak amplitude category 7 list of chords

[11] the multiphonics were chosen using Kientzy's *Les sons multiples das saxophones* (1983)

Naturally, the ocean analyses' lists of chords and the saxophones' multiphonics cannot be exactly matched. In some cases, there is a very fortunate degree of correspondence between a given ocean chord and the saxophone quartet's multiphonic aggregate chord. In other cases, compromises had to be made, such as including a multiphonic that added a foreign pitch to the ocean-derived chord. The following is an example of the process in which the multiphonics were chosen for each saxophone in order to match a given ocean-derived chord (Figure 99). The ocean-derived chord is shown first, under the "Section 32" title, followed by the saxophone multiphonics that include pitches from the ocean-derived chord. Enharmonic microtonal equivalents are common, since in a large number of cases the saxophone multiphonics use flat accidentals, as opposed to the sharp accidentals that characterized the ocean-derived chords. For example, the first saxophone multiphonic in the example below, E𝄳 4 is enharmonically equivalent to the D♯ 4 in the ocean-derived chord.

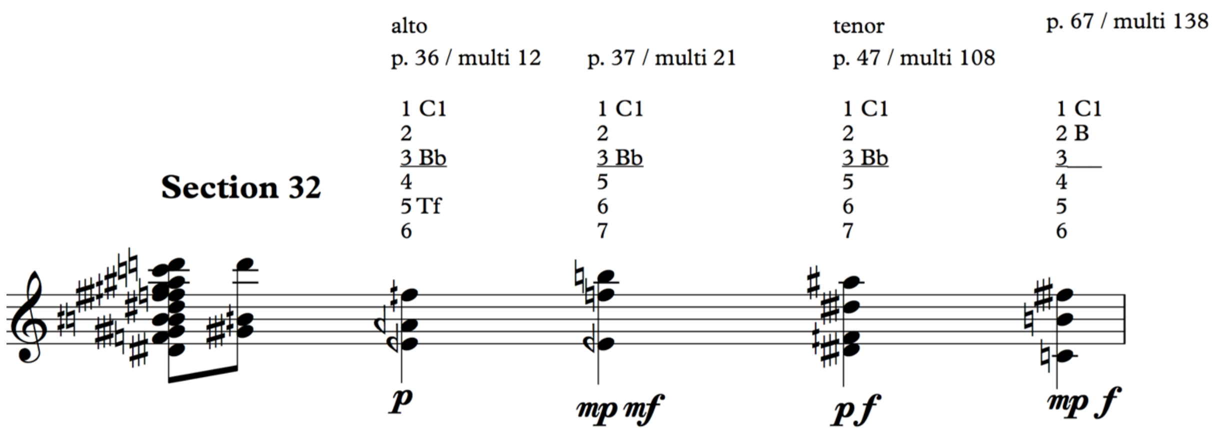


Figure 99. Ocean-derived chord and resulting multiphonics

The last multiphonic in the example above, "multi. 138", for the tenor saxophone, can be seen in the boxed notations in Kientzy's list for tenor saxophone multiphonics. The laborious multiphonic aggregate search suggests how useful it would be to have lists of multiphonics for wind instruments in the XML format. This would facilitate the pre-compositional phases of music that relies in data processing, from environmental sound to algorithmic representations.

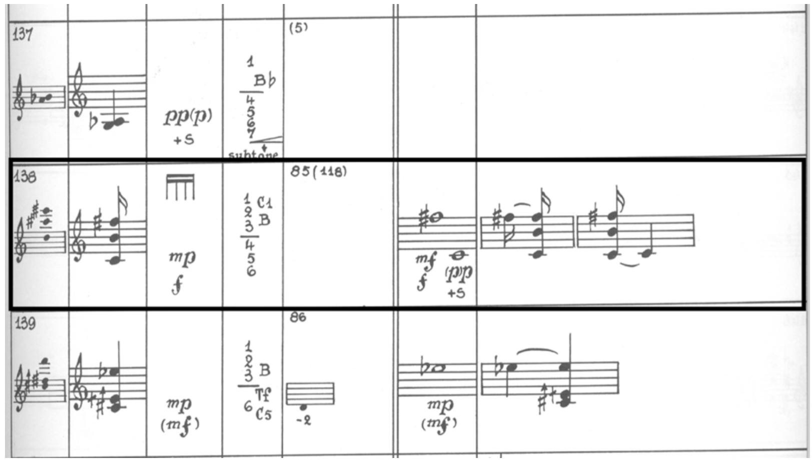


Figure 100. Multiphonic 138 from Kientzy's *Les sons multiples das saxophones*

The following is an example of the aggregate multiphonic section 14 in *Waves Break Aural Shores* (Figure 101). The aggregate multiphonic can be compared to the ocean-derived chord for the same section (instruments are transposed).

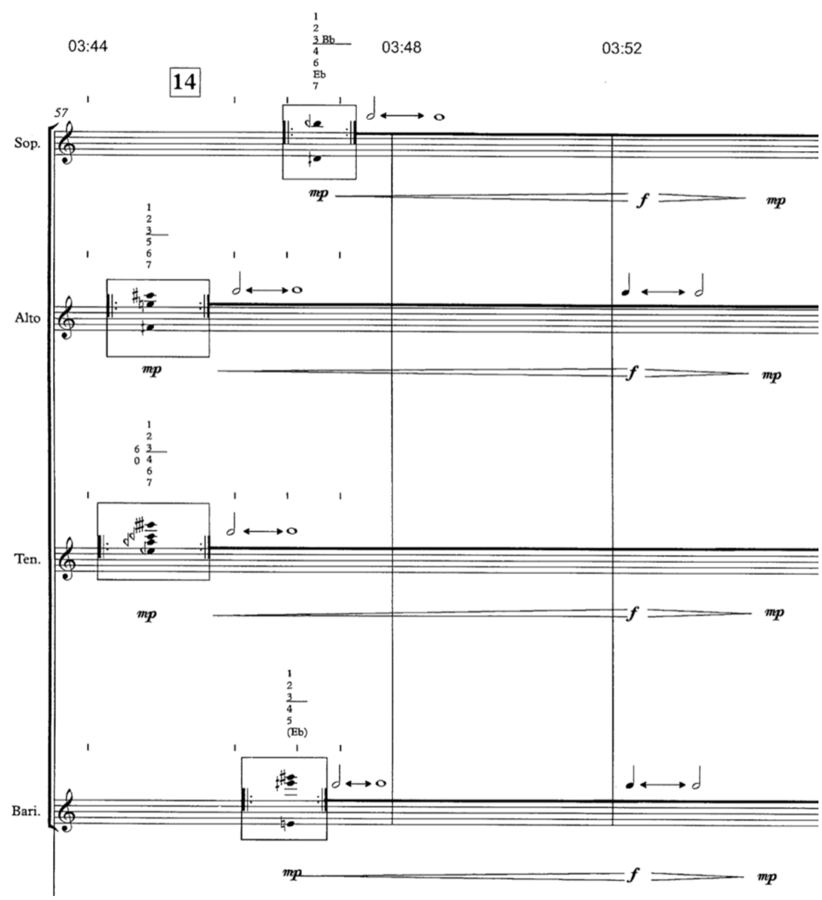


Figure 101. *Waves Break Aural Shores*, section 14, multiphonics

4.1.3.2.5. *Aeolic sound*

This is the only material in the piece that is not derived from FFT analysis. It is the least "objective" material used in the piece due to its lack of connection to the sinusoidal components of the ocean's timbre. Interestingly, however, since the aeolic sound production in the saxophones is characterized by noise, this material assimilates to the ocean's sound more than any other material in the piece. Within the concept and formal plan of the piece, the Aeolic sound material involves a complementary approach in addition to the varying FFT analyses, the varying amplitude threshold analyses, and the loudest average frequency analyses. This material is more intuitive than quantitative and analytical, yet it remains a technique to embody and translate the sound model of the ocean. The technique consists of blowing air into the saxophones with an open embouchure. The parameters to be controlled by the performers during these sections are mainly: 1. dynamics, which are determinate and, as in all other sections, matching the dynamics of the field recording of the ocean; and 2. durations, ad. libitum, according to each performer's breath-holding span and the reed's control of air flow.

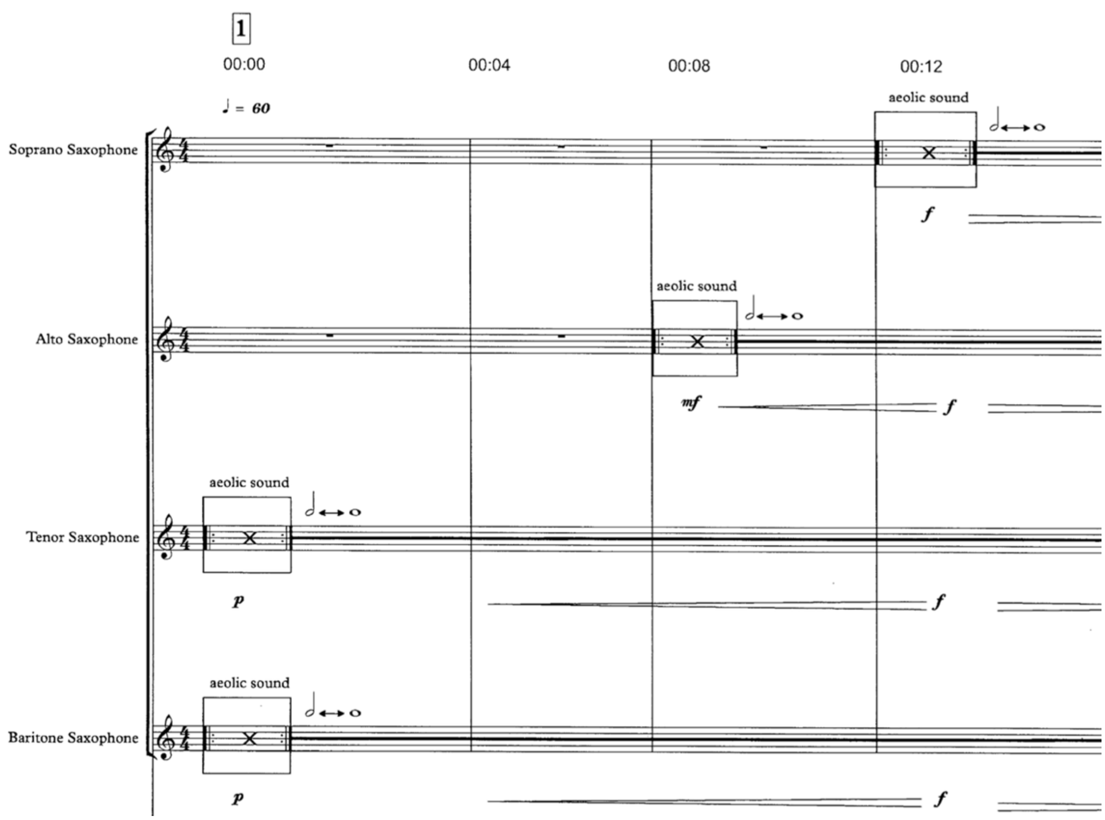


Figure 102. *Waves Break Aural Shores*, section 1, Aeolic sound

4.1.3.2.6. *Dynamics and section durations derived from wave break cycles*

The dynamics and section durations of the saxophone quartet match the dynamics and wave cycles of the ocean recording in the entirety of the piece. After every wave break with an amplitude peak comes a period of quietude that gradually—and sometimes suddenly—builds up to new amplitude peak. Cycles start and end at lowest amplitude before and after peaking. Figure 103, below, shows the first nineteen wave cycles in the recording. The vertical markers divide each wave cycle. The row on the top specifies the cycle number, while the numbers on the waveform specify the amplitude category (1–lowest amplitude; 8–highest amplitude). The row on the bottom indicates the seconds in the recording.

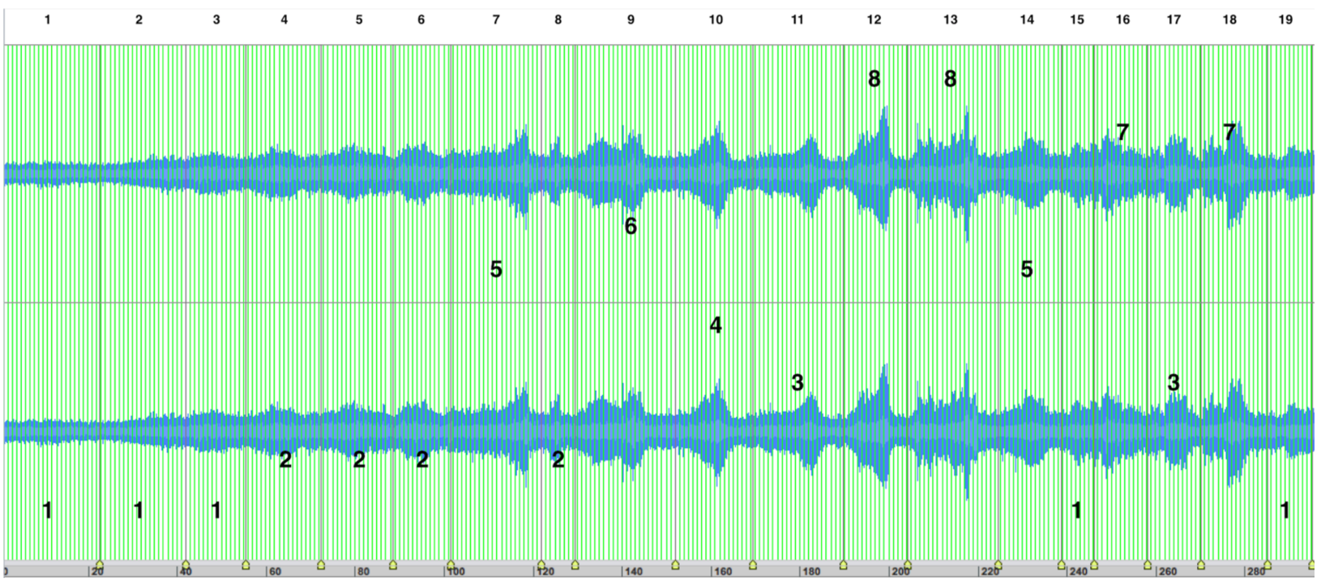


Figure 103. First 19 wave cycles in the ocean recording

Duration proportions of each cycle were precisely transcribed to the musical score by importing the AudioSculpt MIDI markers into the Sibelius engraving software. In Sibelius, the markers are imported as 256th notes (by default), within the context of a quarter note = 60 bpm. Rehearsal marks were written over each marker in Sibelius, labeled with the wave cycle number.

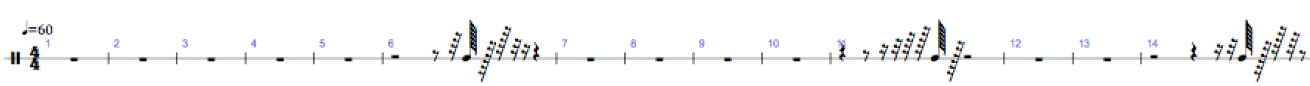


Figure 104. MIDI markers imported to Sibelius

The following example (Figure 105) shows sections 14 and 15 of *Waves Break Aural Shores*. Rehearsal marks 14 and 15 are placed in the exact location in which the AudioSculpt MIDI markers were imported to the score. The durations of wave cycles 14 and 15 were preserved, and the dynamic swells and dips are matching the amplitude contours seen in Fig. n., above, for wave cycles 14 and 15. Lastly, the contrasting material from one section to the other is based on the waves break peak amplitude categories matched to saxophone material (fig. n., section 4.1.3.1.5.). Section 14, peak amplitude category 5, is assigned to multiphonics, while Section 15, peak amplitude category 1, is assigned to aeolic sounds.

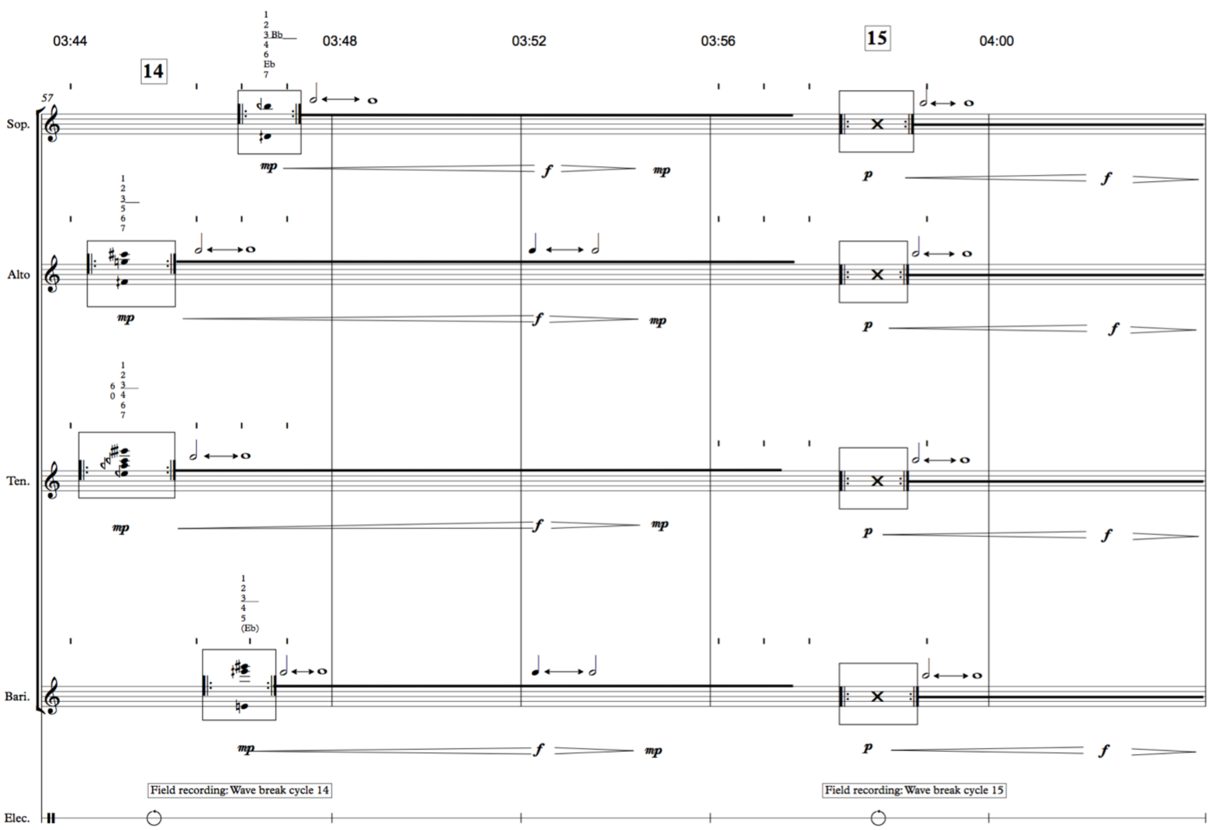


Figure 105. *Waves Break Aural Shores*, score, sections 14 and 15

4.2. Pitched Soundscapes: *Night Music* / *Under the sea ice*

4.2.1. General analytic procedures

Pitched soundscapes are those that include motivic elements that afford recognition due to perceivable fundamental frequencies. (Thoresen, 2007, p. 133) Since environmental soundscapes are often complex due to the simultaneity of phenomena in nature or in anthropogenic contexts, the pitched elements in the soundscapes occur within a more complex environment with higher or lesser degrees of noise. In order to transcribe and analyze the pitched sound models within the soundscapes, the analyst must therefore extract and separate the pitched elements from the noise. This task will greatly vary depending on the soundscape. For example, the sound models in *Under the Sea Ice* are bearded seals recorded underwater in the Arctic. Since there was no sea ice or anthropogenic noise occurring during the time of the recordings, the pitched sounds of the bearded seals could be heard transparently and did not need to be perceptually or spectrally extracted from the noise. On the other hand, the sound models in *Night Music*—crickets and katydids in a Virginia forest during summer—create a collective cacophony that makes the pitch analysis of individual sound models (i.e., a single cricket or katydid) more challenging. Sonograms are extremely useful for this type of soundscape. They provide the visualization of the spectral content of the soundscape, in which it is easy to see foregrounded pitch-defined objects over a background of saturated frequencies or noise.

Cluster timbres and microtonal melodic contours are fascinating attributes of some pitched soundscapes and sound models, such as the two pieces analyzed in this chapter (4.2.2. and 4.2.3.). Sonogram-based manual-aural transcription is essential to recognize the spectral content of some of these cluster timbres. Sonograms can also assist the analyst to estimate the microtonal fluctuation of melodic patterns that considerably exceed most human listening abilities. Once estimations are complete, quantization of the absolute frequencies that constitute the pitched sound-models is the last step before the transcription process. The transcription process implies decisions regarding notation that will influence and are influenced by the compositional idea. This subchapter discusses these general analytic methodologies used in the analysis of two very different sorts of pitched soundscapes and sound models: (1) specific isolated sound models such as the bearded seal recordings and (2) sound models within a broad band cacophony, characteristic of tropical and sub-tropical forests.

4.2.1.1. Sonogram Analysis

Considering the complexity and spectral saturation characteristic of pitched environmental soundscapes, the window size used for the corresponding sonograms should be generally 36,750 samples with a duration of .8333 sec. This is due to the proximity between the sinusoidal components of harmonically complex sound objects in such soundscapes. Considering that “the duration of the analysis window, or time resolution, is inversely proportional to the frequency resolution” (IRCAM, 2011), “the duration needs to be small enough to maintain sufficient resolution between closely adjacent sinusoids, and yet not so large as to average out all of the temporal evolution of the sound’s spectral characteristics.” (McAdams, 2004, p. 175) Addressing this compromise, McAdams suggests a window size “of four to five times the period of the lowest frequency difference between the sinusoidal components of the sound.” The window size of 36,750 samples used for these sonograms is derived from the following generalized information from the complexly pitched biophonic sound models used in *Night Music* and *Under the Sea Ice*:

a) The lowest difference between the sinusoidal components of the sound models (both the stridulating songs of crickets and those of bearded seals) is ~6 Hz.

b) At a 44,100 Hz sampling rate, the lowest frequency difference of 6 Hz corresponds to approximately 7350 samples.

c) A window size of 36,750 samples (5 times 7350 samples) separates the adjacent sinusoids of the recording.

4.2.1.2. Manual-aural transcription in AudioSculpt

AudioSculpt provides a tuning fork tool that plays any selected frequency in a sonogram. In this sense, it functions as a sonogram scanner. As it is clicked and dragged over the sonogram, it plays the sine tone of the frequency over which it is placed and the absolute frequency value is displayed. The transcription process consists of listening to the target sound while viewing its sonogram. Once the sound is internalized in the ear and memory, the tuning fork is used to determine the exact frequencies that constitute the sound or groups of sounds that are highlighted in the sonogram. In 4.2.2.2.3, Figure 119, the main audible pitches in the soundscape (Bb6, D7, F7) are displayed in the sonogram as darker lines that form rows along neighboring frequency regions of Bb6, D7, and F7, covering a wider frequency band than the pitch that they are labeled as. For example, the dark line that includes the Bb6 that dominates the listening is not only that frequency, but a cluster of frequencies between A𝄲 and C#. The same clustering is the case with D7 and F7. This is when listening over the sonogram becomes an essential transcription tool that enables to capture the sound objects that are perceived, specifying predominant frequencies that the sonogram may not specify due to the amounts of information it is able to handle at once.

While the computer is able to manage more data in the transcription, the human listener is able to generate an immediate sonic foreground and background, which is essential for pitched sound models in environmental soundscapes. If the transcription process skipped the listening component, the musical result would be entirely different: the transcribed pitches would not be the Bb6-D7-F7 triad that can be aurally extracted from the soundscape, but three microtonal clusters around each of the pitches of the triad. The transcription would reflect the computer software’s display rather than what the composer hears and wants to hear according to his/her relationship to the sound models. Such transcription would not be a worse or better path, but purely a different conceptual and aesthetic approach. My approach to pitched sound models and soundscapes necessitates the listening component (the manual-aural transcription) because it is consistent with the musical world that I am listening from. In this sense, when environmental sounds allow musical recognition and translation, my priority is to listen to them, internalize them, and embody them by composing with them. This is perhaps the ultimate reason why I tend to find the sonogram-aided manual-aural approach more fitting than automatic transcriptions using partial tracking.

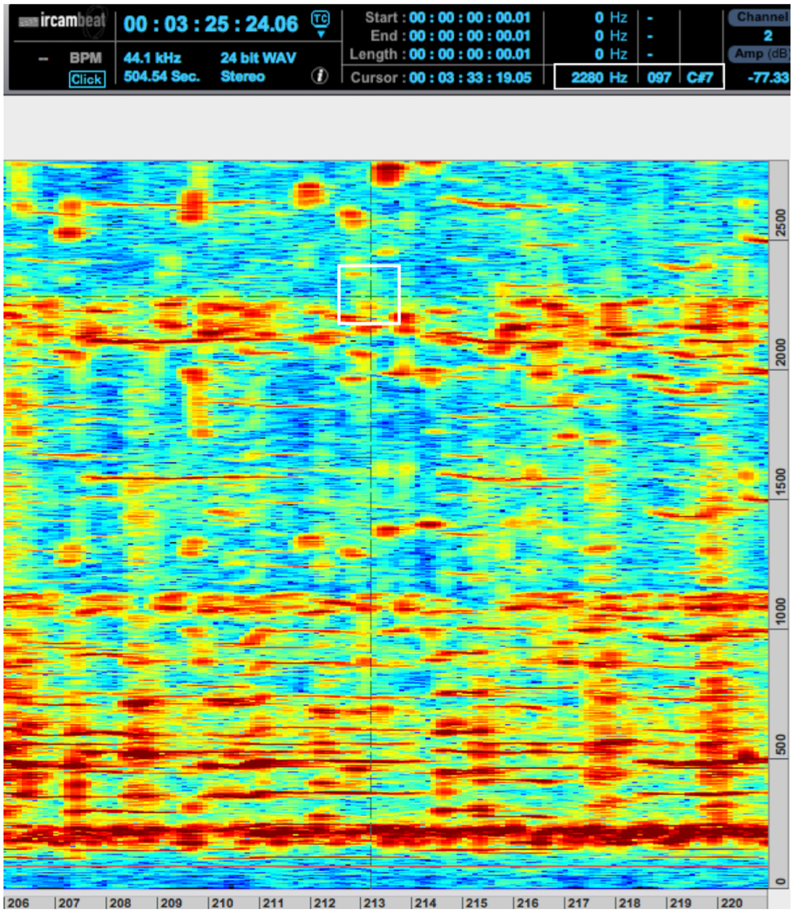


Figure 106. AudioSculpt tuning fork function (selected frequency target in white square on the sonogram; frequency information in white square above the sonogram)

4.2.1.3. Manual-aural transcription in SPEAR

Similarly to the process aided by AudioSculpt, the transcription is done by listening to the audio while examining the sonogram. Instead of using the AudioSculpt tuning fork sinusoidal identifier tool, SPEAR allows to highlight specific re-synthesized sinusoids that more precisely match the fundamental frequency heard at specific points in time. This feature is useful for long, continuous pitched sounds (i.e. 5 seconds to minutes of duration) with varying frequencies, since it allows to examine a number of highlighted varying fundamental sinusoids that otherwise would be impossible to retain in memory. This method was used for the transcriptions of the six bearded seal calls that constitute the material for *Under the Sea Ice*, for string quartet and electronics.

In the example below (Figure 107), bearded seal call “AL1i”, an initial ascending motion is followed by a long descending line of smaller ascending and descending glissandi. The selected re-synthesized sinusoids in red represent the fundamental frequencies heard between seconds 6 and 9 of the sound model’s re-synthesized audio file. In the bottom left of the figure, the Cursor displays the frequency (1680 Hz) of the first selected sinusoid at 00:06; the note display in the cursor displays the absolute note and register at G#6; the Keynumber displays the Midi value of the frequency (92.195), in which the decimal denotes the microtonal increase of a rounded 1/8 of a tone. Since the transcription of some of this information to music notation takes some time, it would be impossible to retain the sound and values of more than two or three sinusoids. In this sense, highlighting allows to cover a broader timeframe, which is relatively more time efficient in the extremely time consuming task of transcription.

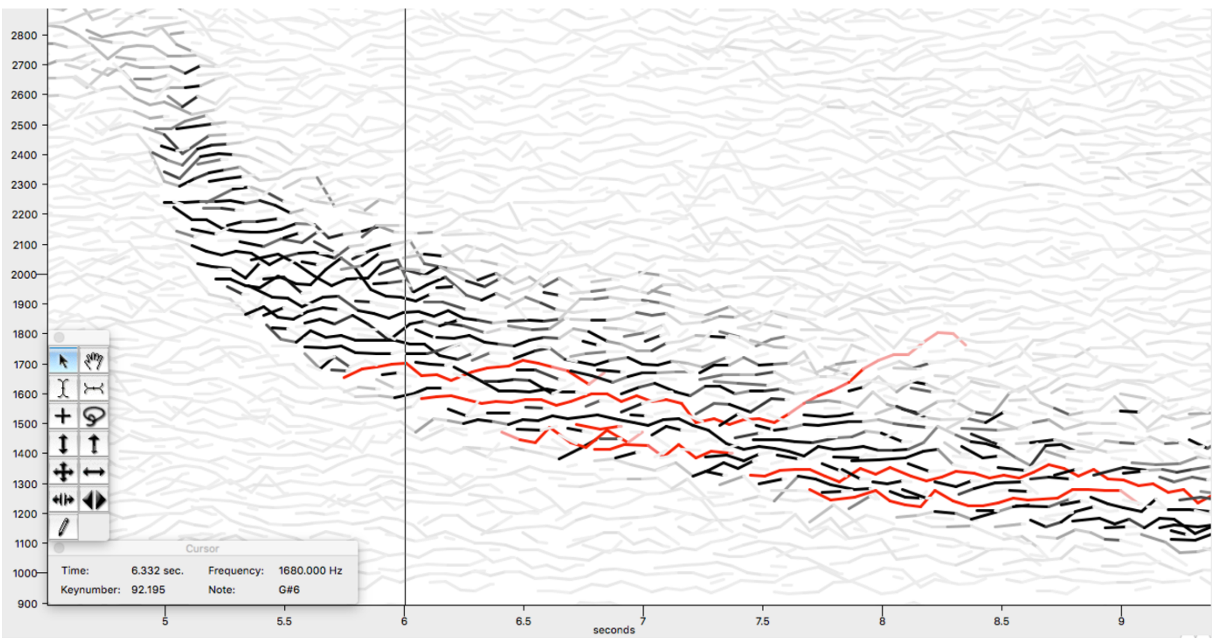


Figure 107. Bearded seal call “AL1i”, selected sinusoids in SPEAR

The example below (Figure 108) is the transcription of the first eleven seconds of the“AL1i” sound model, in which time is indicated in seconds, placed above the staff. For practical transcription and performance purposes, notation is proportional.

Figure 108. Bearded seal call "AL1i", manuscript of the transcription

4.2.1.4. Frequency to quantized pitch

Once the exact frequency of the sound model is detected, it is translated into midi float (decimal in Max language) numbers. In the context of midi numbers, which are absolute numbers assigned to absolute note values (i.e. midi 46 = Bb6; midi 47 = B6; etc.), the decimal adds the fraction of the tone equivalent to the fraction of the tone being added. For example, 47.5 = B ‡ (1/4 tone sharp). In Max, the absolute frequency number displayed is sent to a "ftom" (frequency to midi) object. The quantization is performed by the "round" object set to a value that reflects the desired microtonal resolution. Since midi note numbers are assigned to consecutive semi-tones (midi 60 = C4; midi 61 = C#4; etc.), "1" is assigned to a semi-tone, "0.5" rounds to a quarter-tone, "0.25" rounds to an eighth-tone, and so on. The following is an example of a frequency value converted to a midi note number, quantized in a resolution of an eight of a tone (Figure 109):

2539 Hz ———— 99.34 midi note ———— 99.25 rounded value ———— E 5/8 tone flat

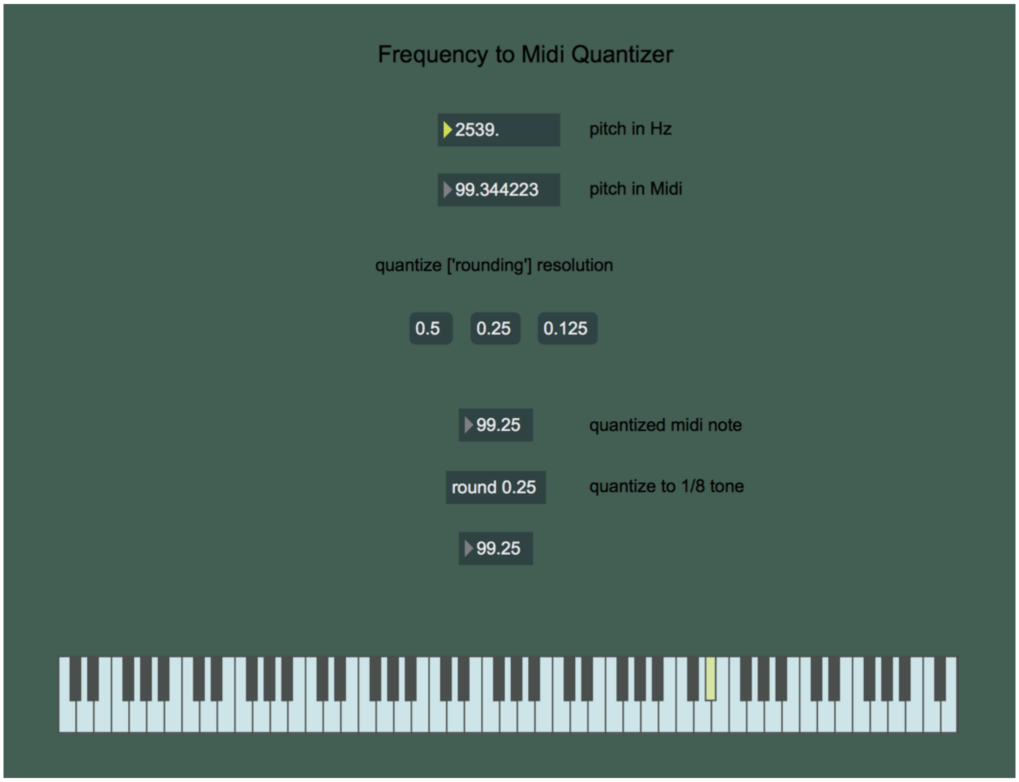


Figure 109. Frequency value converted to a midi note number, quantized in a resolution of an eight of a tone (Coffey, n.d.)

4.2.1.5. Proportionally notated transcription

The choice of proportional notation (spatially distributed) for the transcription of the pitched sound models presented in this chapter responds to an aesthetic-philosophical reason and a practical issue. From a philosophical perspective, environmental natural phenomena are never repeated exactly. As Heraclitus observed, "No man ever steps in the same river twice." No cricket stridulating phrase, wave cycle, or rain pattern repeats in the exact same way. The exactitude of determinate notation in music guarantees a more or less exact repetition of the notation in subsequent performances. On the other hand, proportional notation allows to be temporally precise in the descriptive score, while at the same frees the performer from the temporal grid nested in rhythm and allows interpretation of exactly where the sound will be placed in time. The result is temporal fluidity and a higher degree of variance from performance to performance, while still preserving the sequence of temporal events in the environmental soundscape.

From a practical perspective, the temporal complexities of chaotic events imply a high degree rhythmic complexity in a determinately notated score. In my own compositional approach, there is no practical reason for having a sound played exactly at the fourth subdivision of a nonuplet (Figure 110a) instead of at the fifth subdivision of a nonuplet (Figure 110b). The notational nuance for such precision throughout the entire score would require a significant amount of the performer’s time, which is much better utilized in listening to the sound models that originated the notations and embodying them in every sound emitted. Proportional notation (Figure 110c) reflects the sound models’ rhythmic content with an adequate degree of temporal detail, closely resembling the results of rhythmically precise determinate notation.

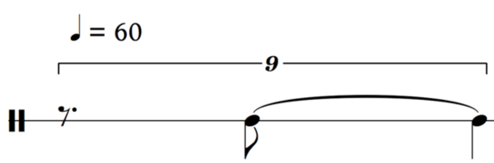


Figure 110a determinate notation example

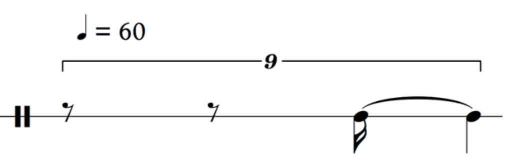


Figure 110b determinate notation example

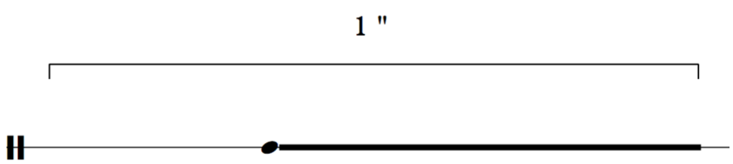


Figure 110c, proportional notation example

It is important to mention that the sound models should suggest not only the type of notation (determinate, indeterminate, proportional, etc.), but the general analytic procedures as a whole. While the two pieces presented in the Pitched Soundscapes portion of this chapter necessitated proportional notation, other pieces derived from pitched soundscape sound models may work well with either determinate notation or other forms of notation (indeterminate, textual, etc.). Most importantly, the ears and imagination should always dictate how to best translate the environmental soundscape into music.

4.2.2. Night Music: Frequency, Dynamics, and Density Content of Stridulating Insects in a Spatialized Field Recording

Night music is a piece for reed quintet (oboe, Bb clarinet, alto saxophone, bass clarinet, and bassoon) derived from direct transcriptions and arrangements of a 5-channel recording of the summer dusk and night sounds of stridulating insects in a Virginia forest. The piece is structured in five movements, each taken from a fragment of the 40-minute original recording. The striking increase in harmonic density and loudness as the recording unfolds is the guiding formal principle of the piece. The recordings, featured in the electronics, were made with five simultaneous microphones in a pentagonal distribution, at a distance of ~30 meters between each mic. Each microphone analysis and transcription was assigned to an instrument. The multi-channel recording sought an expanded listening field resulting from the different microphone responses and placings.

4.2.2.1 Field Recording

4.2.2.1.1. Background Noise and Location

The recording was realized in the Walnut Creek Park, situated in the North Garden unincorporated community in Albemarle County, Virginia. The location was decided based on data obtained from the Virginia Department of Transportation's Annual Average Daily Traffic information (Virginia Roads, 2019). This document provided useful information to avoid areas with high noise pollution derived from ground and air transportation. The average daily traffic (shown in bold font in the Figure 111) in the nearest roads to the park is 1,300, of which 65% of the traffic volume travels in the peak hour. In comparison, other parks within this area of Virginia are situated near roads with an ADT of 52,000. For field recordings that will provide sound models for composition, this consideration is essential.

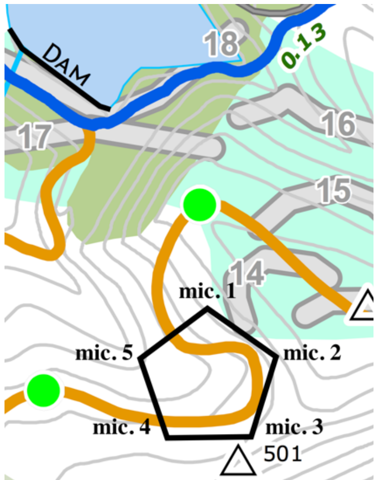
631	Old Lynchburg Rd	From: 02-712 E. Plank Rd	To: 02-708 Red Hill Rd	3.15	**800**	G	97%	1%	1%	1%	0%	0%	F	0.140	0.613	830	G	2016
631	Old Lynchburg Rd	From: 02-708 Red Hill Rd	To: 02-706 Dudley Mountain Rd	3.92	**1300**	G	97%	1%	1%	1%	0%	0%	C	0.102	0.651	1400	G	2016
631	Old Lynchburg Rd	From: 02-706 Dudley Mountain Rd	To: 02-875 Country Green Rd	1.98	**2300**	G	97%	1%	1%	1%	0%	0%	F	0.1	0.657	2400	G	2016

Figure 111. Annual Average Daily Traffic Chart (Virginia Department of Transportation)

4.2.2.1.2. Spatialization: Expanded auditory and perceptual fields

Five rigs with microphones of various specifications were placed around an imaginary pentagon of ~30 meter sides (Figure 112). Varying microphones were used instead of matched microphones in order to compositionally explore the variances between the sound perceived by each microphone. Such perceptual variances consist of subtle differences in pitch which determine the harmonic content of the piece.

There were two objectives in the realization of the spatialized field recordings and the resulting sound model-derived piece: (1) Capturing the constantly shifting sound in the forest throughout the recorded perimeter. A by-product of this goal was an expanded auditory field, comparable to a 360º photo, where the perceived sonic events are distributed throughout the perimeter. (2) Continuing research on the various modes of perception and their compositional implications, which has been present both in my creative work (i.e., *Waves Break Aural Shores*, for saxophone quartet and electronics, in 4.1.3) and my analytic work (i.e., *Peter Ablinger's 'Quadraturen IV', perception and reproduction of environmental sound through instrumental music"*, presented in 3.2). *Night music* explores the different contents that each microphone and recording device captured of the same environment. These varying perceptions integrated within the same system result in an expanded perceptual field. The spatial distribution of the field recording is recreated in the performer setup for the live performance of the piece. The performers surround the audience in an imaginary pentagon.



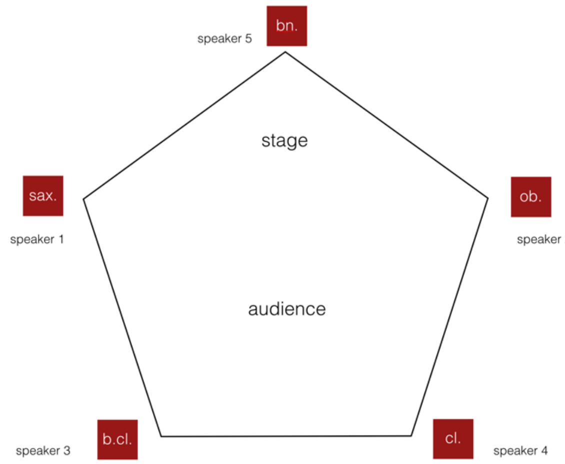


Figure 112. Map of Walnut Creek Park microphone setup location (each side of the pentagon is ~30 meters), and corresponding concert loudspeaker configuration.

4.2.2.1.3. Microphones

The following microphones were used, considering their frequency response suitable for the stridulating insects:

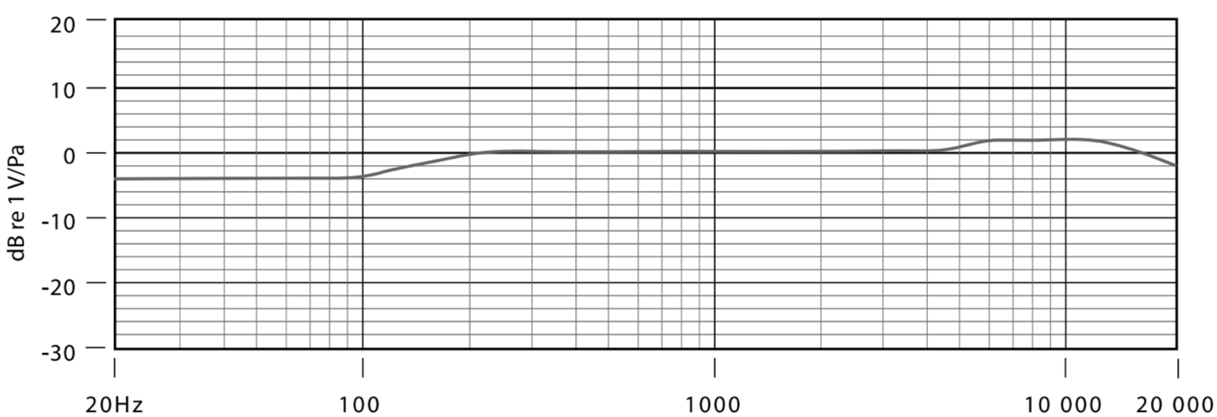


Figure 113. Mic. 1–Cardioid Condenser Rode NT4 (Stereo)

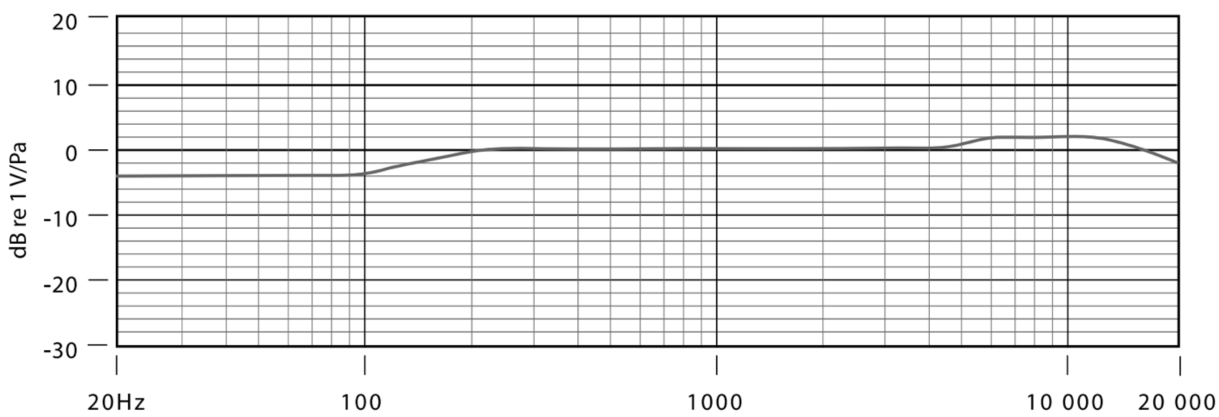


Figure 114. Mics. 2, 3, and 4–Cardioid Condenser Rode NT5 and MBP 603 (Mono)

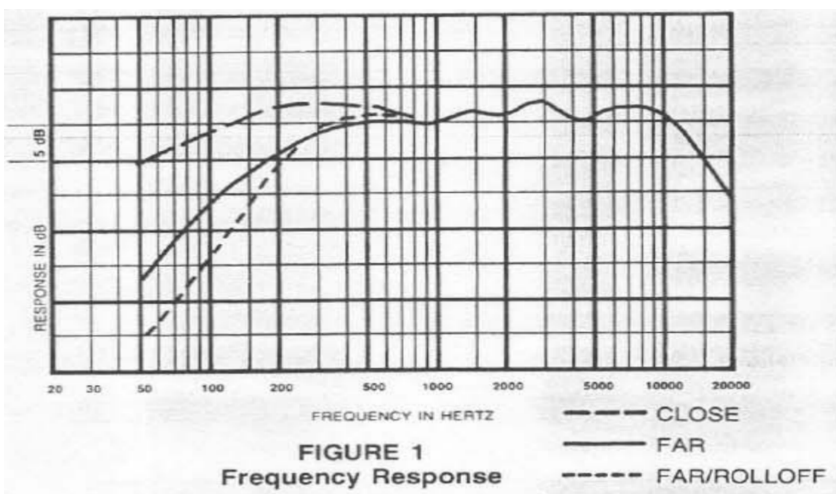


Figure 115. Mic. 5–Cardioid-line Dynamic Shotgun Electro-Voice (Mono)

4.2.2.2. Sound Models

There are several layers in the summer dusk recording. A selection of these were used as sound models that were transcribed, analyzed and orchestrated for the reed quintet. The sound models are the following:

4.2.2.2.1. *Nature's Gift of the Unexpected—Katydids' Pulses* Versus *Cicadas'* Continuum

The original idea was to record the continuum of the sounds of cicadas sweeping through the pentagonal microphone setup, capturing the spatialized movement of the sounds' chorography in the forest. However, due weather conditions, the cicadas' sound was absent, and the characteristic rhythmic pulses of the Common True Katydids, commonly known as "katydids", gradually dominated the soundscape in a 20 minute build-up. On the day of the scheduled recording, the temperature was approximately 15° F cooler than predicted and the humidity dropped considerably. These fluctuations resulted in the silencing of the cicadas. Instead of the expected sound model, the sound of the katydids consisted of groups of three to four brief and loud broadband pulses per second which, multiplied and scattered throughout the perimeter, produced a cacophonous sonic glitter that provided the direction and form of the piece.

In the images below (Figures 116a and b), the cicadas' continuum produced in higher and more humid temperatures is compared with the katydid's brief rhythmic bursts that became the sound model. The different iterations of the 3-4 pulse sets are presented simultaneously in the five channels of recording. This multiplicity in the periodicity of each of the channels results in an effective material for a spatial/surround projection of the sound with the reed quintet surrounding the audience.

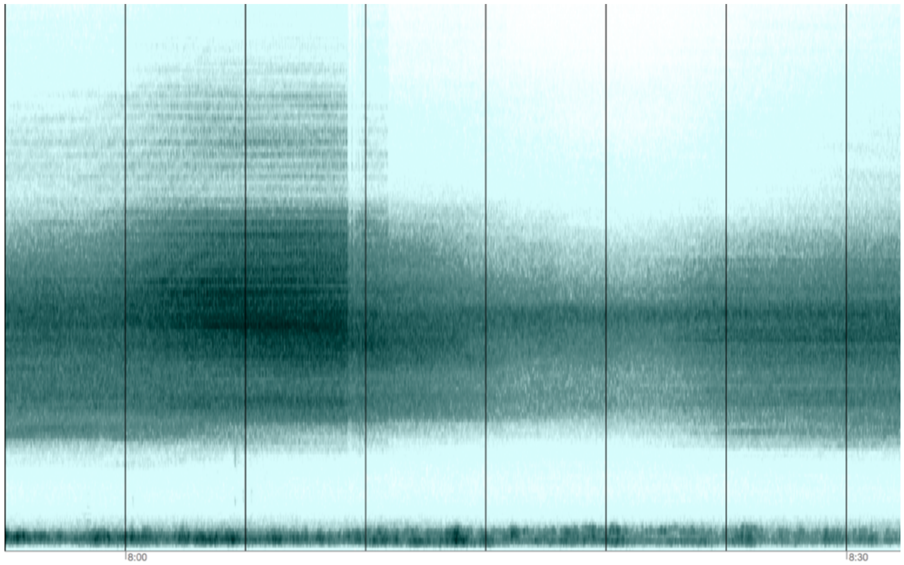


Figure 116a. Cicadas' continuum in 1 channel

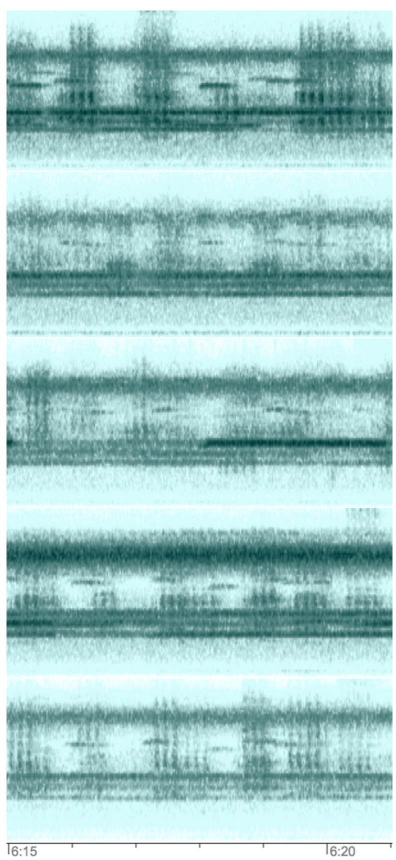


Figure 116b. Groups of 3 to 4 katydid pulses in 5 channels

4.2.2.2.2. Katydid Pulses Gradual Saturation

One of the most striking features of the soundscape of stridulating insects and cicadas in hot climate seasons is the gradual increase of sound activity that takes place in dusk. The density and loudness of the recording changes dramatically from its beginning (7:30 pm) to its end (8:10 pm). The way in which the gradual saturation takes place in the course of 40 minutes is seamless, without discontinuities in intensity, always filling the acoustic space and time by means of short increments.

From a sound-model perspective—translating the sonic parameters of this dusk soundscape into a composition—, the gradual increase of acoustic energy seemed like a promising formal concept. Regardless of the fact that the formal direction itself is not new—various pieces from the last two centuries display a gradual increase in energy towards a climactic ending—, the manner in which a limited number of relatively simple musical materials accumulate so gradually in a multiplicity of periodicities elicits relevant experiments in the temporal domain.

In the examples below (Figures 117a–c), varying degrees of saturation of the sets of katydid pulses are shown in three different successive moments in time. In the first of the examples, the black vertical brackets highlight the iterations of the pulses. Note the contrasting spacing of the pulses between the beginning 30 seconds (from 1220 to 1250) and the last 20 seconds (1300 to 1320) of the first example (Figure 117a).

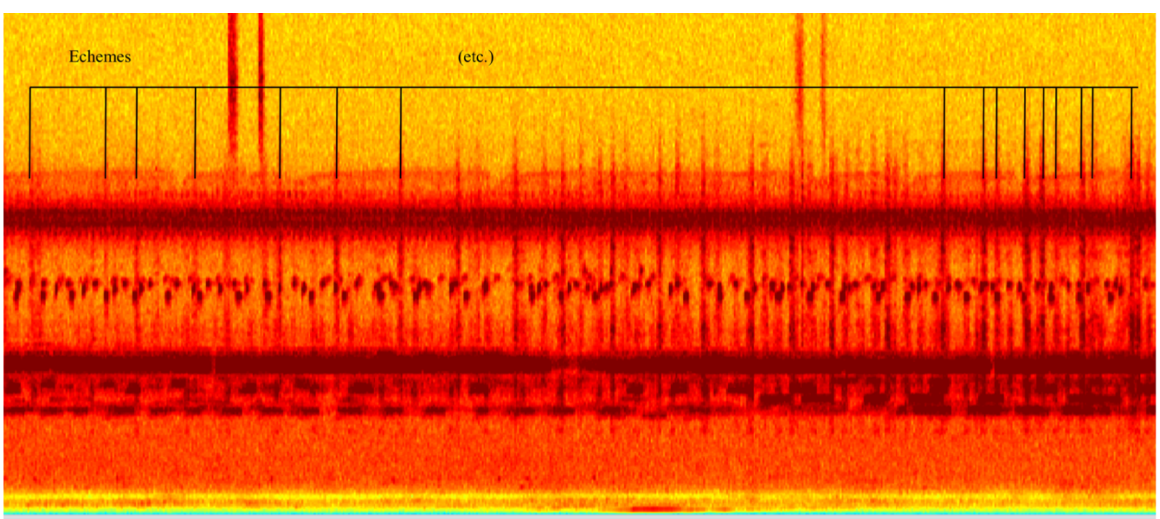


Figure 117a. Gradual saturation of pulses (low saturation)

In the following two examples (Figures 117b and c), the saturation is considerably higher, not only in the rate of repetition of the pulses, but also across the frequency bands.

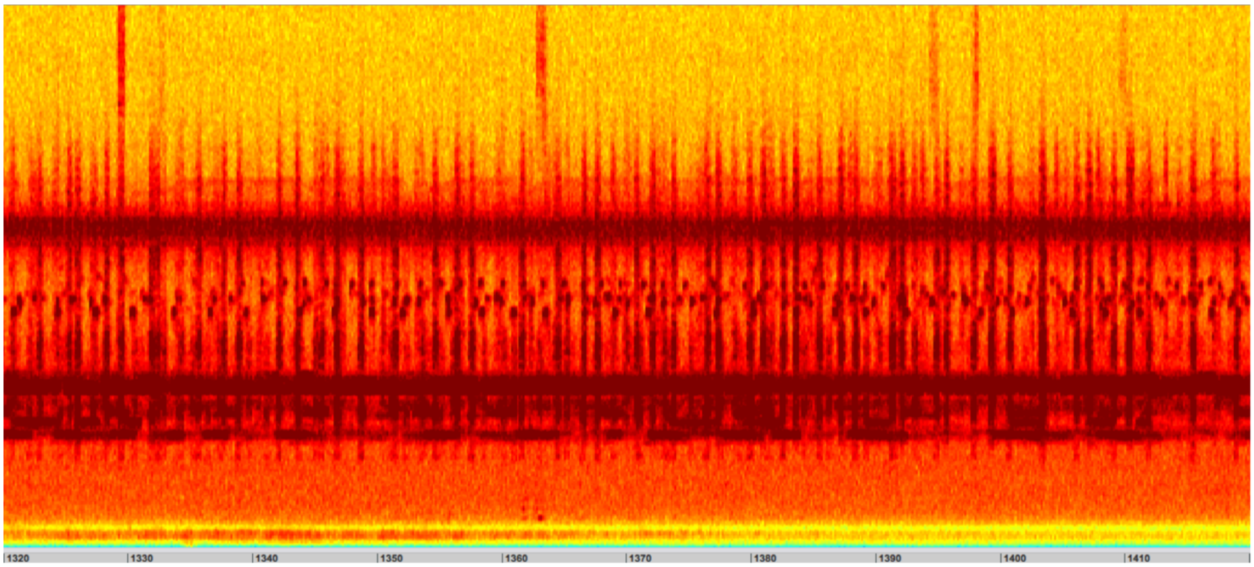


Figure 117b. Gradual saturation of pulses (medium saturation)

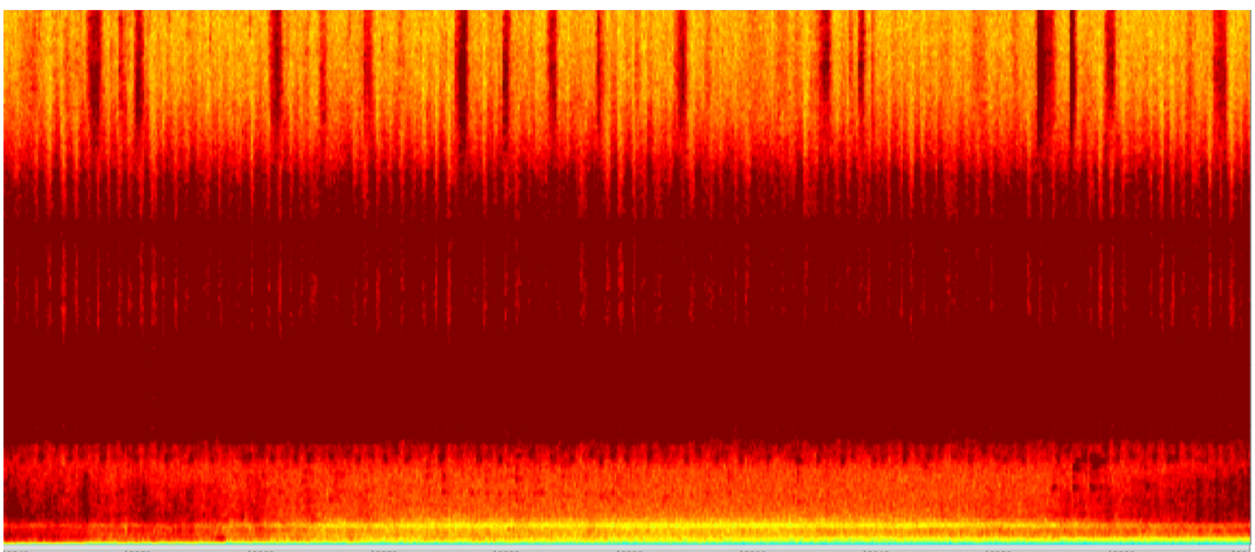


Figure 117c. Gradual saturation of pulses (high saturation)

4.2.2.2.3. Stridulating Tunings (Cricket Chorus)

Another important element of the field recording is the cricket stridulation, producing micro-tonal clusters around each of the pitches (Bb, D, F) that form the Bb major triad in root position. The complexity of the multiple iterations of sounds tuned at and slightly around the Bb major triad, makes this material one of the two most explored sound models extracted from the field recording.

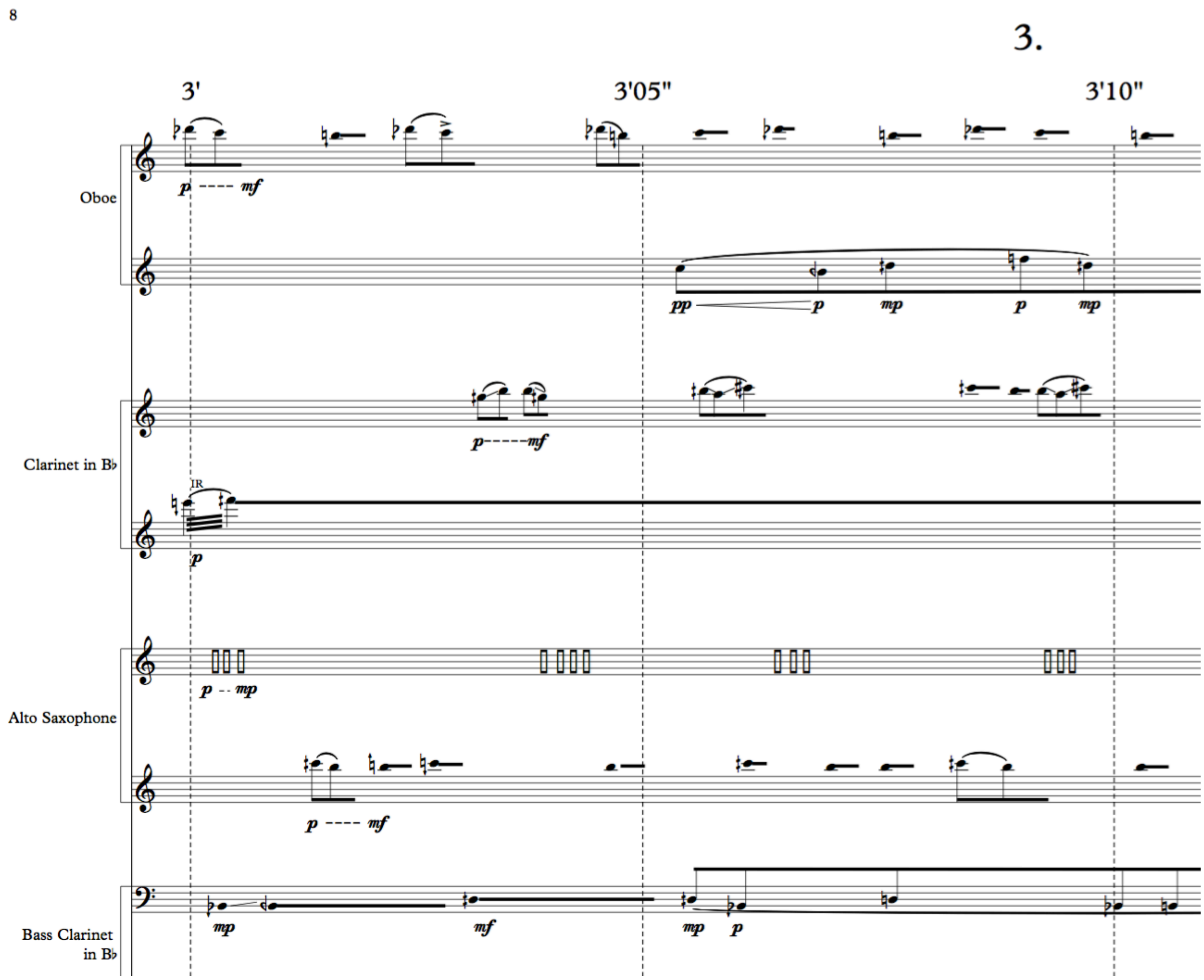


Figure 118.
Microtonal fluctuations around the Bb major triad pitches in Ob. (bottom staff) and Bass Cl. (upper staff)

In the image below (Figure 119), the three pitches in the Bb triad in the crickets' stridulation's original register—the 6th octave—are displayed. The thickness of the lines show the clustering of partials around each of the pitches.

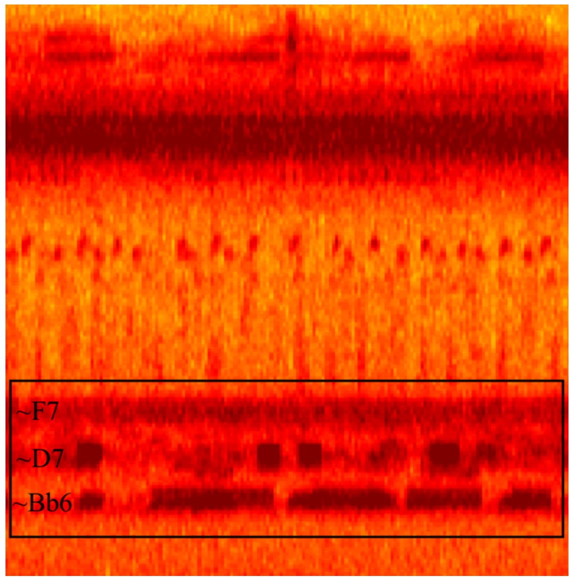


Figure 119. Micro-tonal clusters forming a Bb major triad in the 6th octave

From a modal perspective, the pitches derived from the stridulating crickets created a constantly changing and superimposed set of modes around Bb6 and its microtonal tonic iterations (microtones around Bb6). Throughout the 48 minutes of recording there is a prominent Bb6 (plus its microtonal neighbors) drone resulting from a mass of stridulation that becomes emphasized as the recording progresses. In addition to that drone, other cricket phrases are produced at frequencies that, grouped together, may suggest Ionian, Lydian, and Mixolydian modes with some tuning differences.

4.2.2.2.4. Microphones' Perceptual Variations of the Same Source

As mentioned in the *expanded auditory and perceptual fields* (4.2.2.1.2.) section above, one of the experiments of this piece was to examine the perceptual variations derived from different microphones spatially situated. The cricket chorus provided useful material for this task, given the pitch specificity of each of the distinct crickets' stridulations. A striking finding in the spectrum/sonogram analyses of each of the 5 microphone recordings was that when two or more microphones presented the same sonic event, there were subtle variations in the tuning of the event. The two main factors for these variations are: 1) location and 2) frequency response. With regards to the former, considering that a space behaves like a filter that amplifies and attenuates certain frequencies, a specific forest location filters the sound differently than another forest location. In the case of the latter, each microphone picks up a different range of frequencies, which will yield subtle differences when one sound is recorded by two or more different microphones.

The sonograms below (Figure 120a and b) present the same sound model fragment in two different microphones (mics. 4 and 5). The stridulation analyzed is displayed by a rectangle. The horizontal line displays the frequency with the highest amplitude in the stridulation, which determines the perceived pitch. The loudest frequencies in both microphones differ by 40 Hz, which in that frequency band results in a distance of ~1/8 of a tone. This is a considerably subtle difference in the pitch content of each microphone. Used as a sound model for instrumental material, these micro-variations resulted in a micro-polyphony derived from the same sound object.

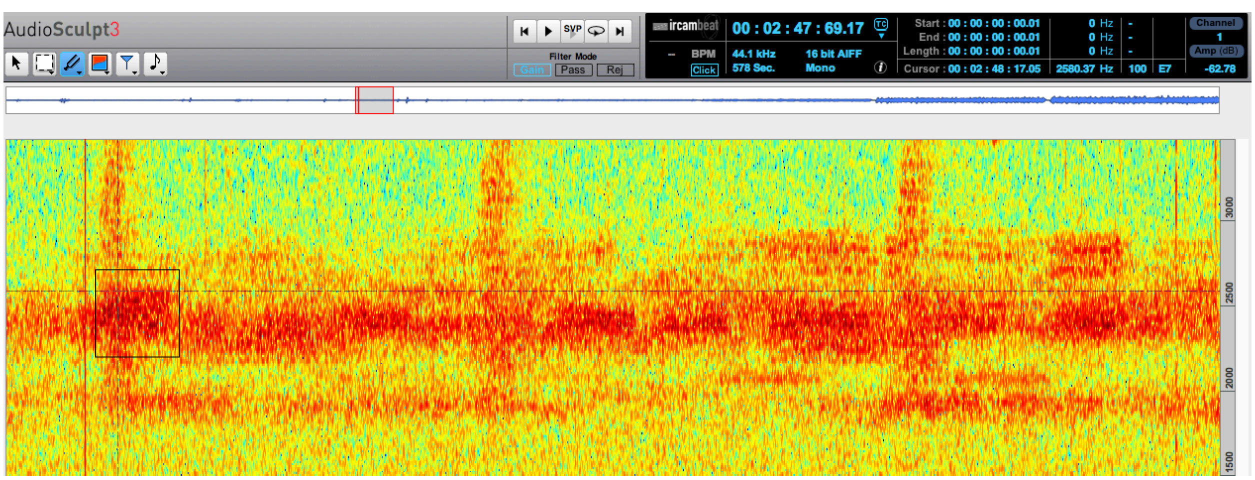


Figure 120a. Microphone 4, cricket stridulation at 2580 Hz (rounded to D♯)

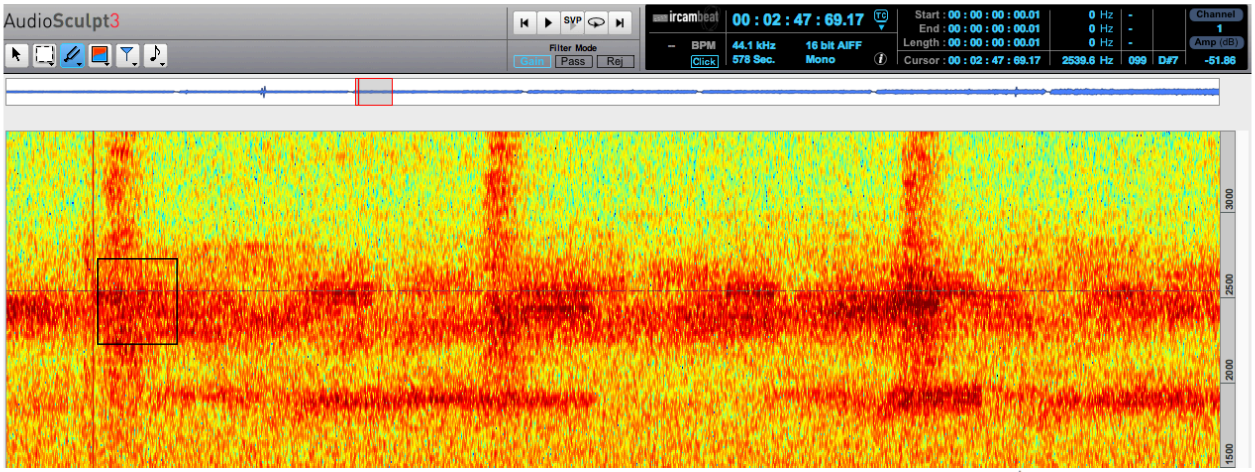


Figure 120b. Microphone 5, cricket stridulation at 2539 Hz (rounded to D♯)

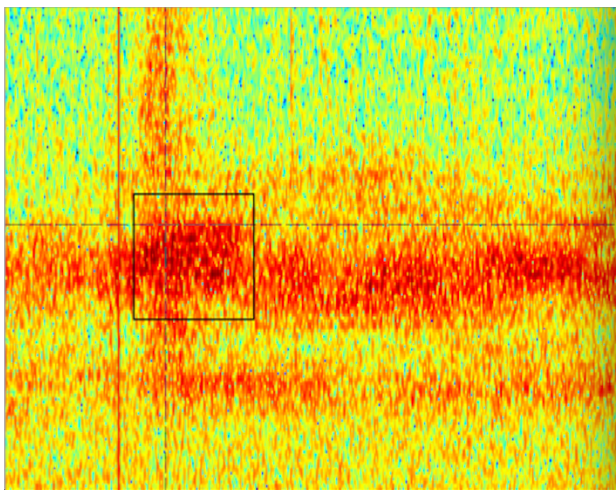


Figure 120a. 2580 hz | (rounded to D♯)

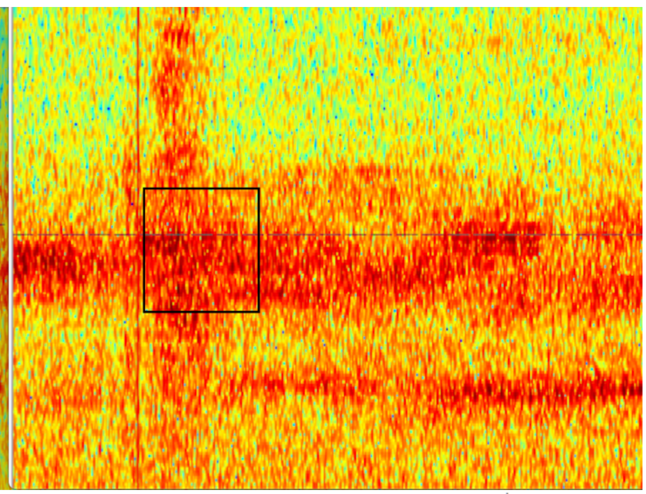


Figure 120b. 2539 hz | (rounded to D♯)

The expanded perceptual fields generated by the microphones' differences suggested a compositional idea in which counterpoint and timbre become fused. The transcribed and instrumentally re-assigned material from each microphone, with its minimal pitch differences, yields a micropolyphony in which the distinct pitches of each instrument occasionally merge as a single timbre when they share the same register. In the example below (Figure 121), two different timbres are derived from the polyphony: 1) the micro-clusters around Bb in the oboe, bass clarinet and bassoon (red boxes); 2) the micro-clusters around F in the clarinet, alto sax and bassoon (blue boxes).

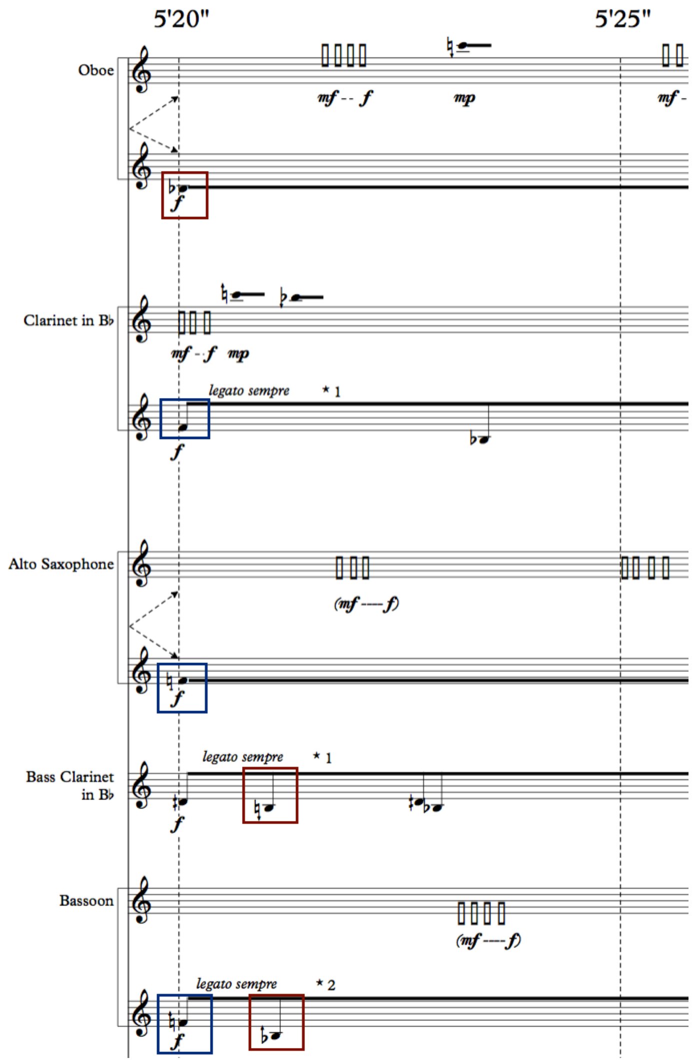


Figure 121. *Night Music*, micro-clusters around Bb and F

4.2.2.3. Compositional / Analytical Approach

The distinct sound objects in the summer dusk soundscape, many of which present a relatively transparent pitch content as well as clearly defined motivic traits, were the deciding factor to work with manual-aural transcription for their translation into musical material. Partial tracking was tested in the pre-compositional stages of the piece, yielding results that featured more rhythmic complexity and pitch information due to the subtle microtonal variations within single sounds (Figure 122a). On the other hand, manual-aural transcription generated less rhythmic complexity and pitch information due to how the human perceptual apparatus tends to group features into one entity during the transcription process (Figure 122b). While the partial tracking process presented a 1.2” stridulation as a trill of 22 pitches around D#, the manual-aural transcription method aided by pitch-detection software derived in a single D♯ pitch that grouped all the slight pitches around it. Considering that the microtonal tunings and the transparent harmonies derived from their simultaneities were one of the most striking features of the recordings, I opted for the harmonic simplicity of a manual-aural transcription aided by sonographic information in AudioSculpt. Another factor that informed the choice was the feasibility of performance given the rehearsal time for the concert and recording session. The partial tracking transcription yields hyper-complex material which, although quite promising, entails exponentially increased hours of rehearsal time. The examples below (Figures 122a and 122b) show the same sound model transcribed with the two different methods: 1. Partial tracking; 2. Manual-aural transcription.

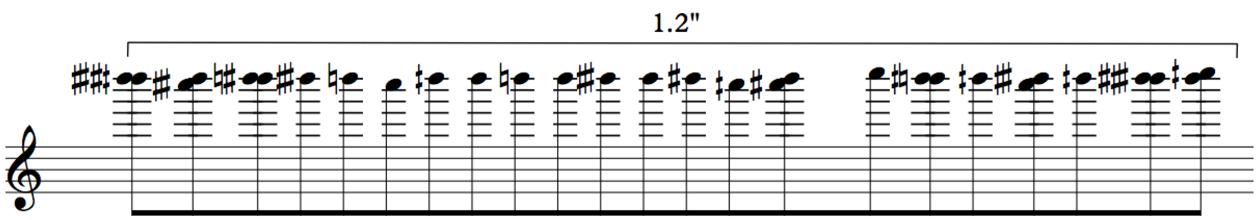


Figure 122a. Microphone 4, cricket stridulation. Quantized sdif file in OM after partial tracking

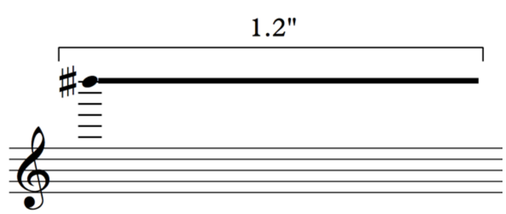


Figure 122b. Microphone 4, cricket stridulation.
Manual-aural transcription (proportional notation) using AudioSculpt sonogram and tuning fork tools

4.2.2.3.1. Partial Tracking

The translation of partial tracking of cricket stridulation (Figure 124c, below) into music notation (Figure 122a, in 4.2.2.3) provided detailed timbral information regarding frequency content. Despite the partial tracking method was not used for the final transcription of the field recordings due to the aforementioned reasons in 4.2.2.3, it provided important insight that informed notation and performance expression –phrasing, timbre, and articulation, mainly. For example, in movement 3, the bass clarinet plays an irregular tremolo including all the pitches between D2 and F♮2, as rhythmically saturated as possible (Figure 123) This notation is an abstraction of the detailed and hypercomplex transcription that results from the partial tracking process, as shown above (Figure 122a, in 4.2.2.3).

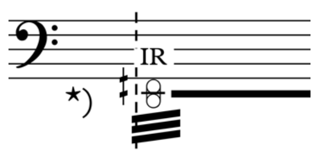


Figure 123. *Night Music*, bass clarinet irregular tremolo section

Similarly to the partial elimination process used in *Waves Break Aural Shores* (4.1.3), the first part of the analysis and transcription process consisted re-synthesizing the audio files in SPEAR. After listening for the target cricket stridulations in the re-synthesized files, all other partials and frequency bands were deleted (Figures 124a–c).

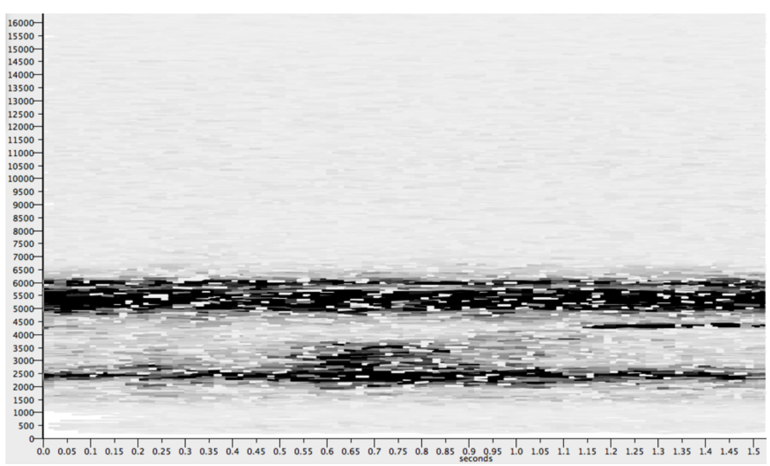


Figure 124a–c. Microphone 5. Cricket stridulation re-synthesis; gradual filtering of frequency bands until only the re-synthesis of the target cricket's pitch remains. Figure 124a. No partial elimination

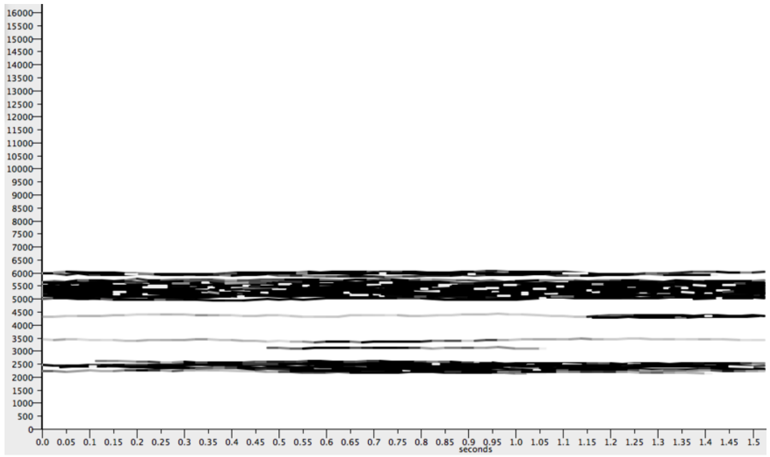


Figure 124b. Elimination of partials under a given amplitude threshold

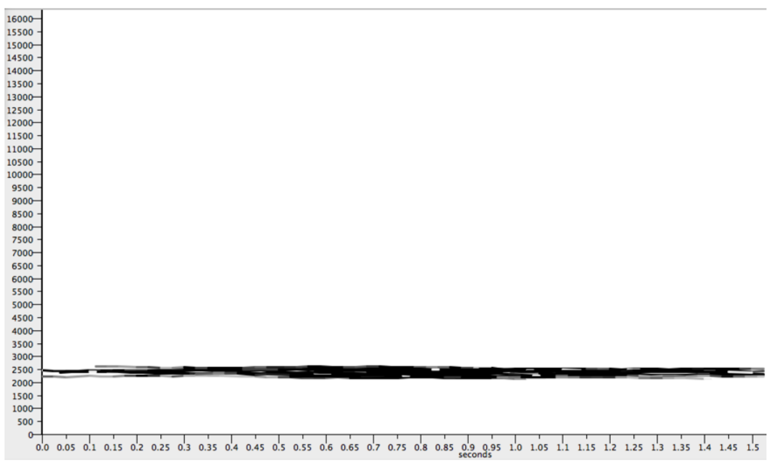


Figure 124c. Elimination of all partials except for cricket's manual-aurally detected pitch

Once the re-synthesized SPEAR file kept only the cricket stridulating sounds, the .sdif file was loaded in an OpenMusic patch (Figure 125). The main function of the patch is to translate the re-synthesized .sdif file into music notation with quantization in the time and frequency domains. The main object used in the patch is the "voice" object, which takes an .sdif file as input and generates a sequence of chords with a rhythmic structure. The "voice" object allows varying degrees of time and frequency resolution, and exports its output to MusicXML, which is the standard open format used in music engraving software such as Sibelius and Finale.

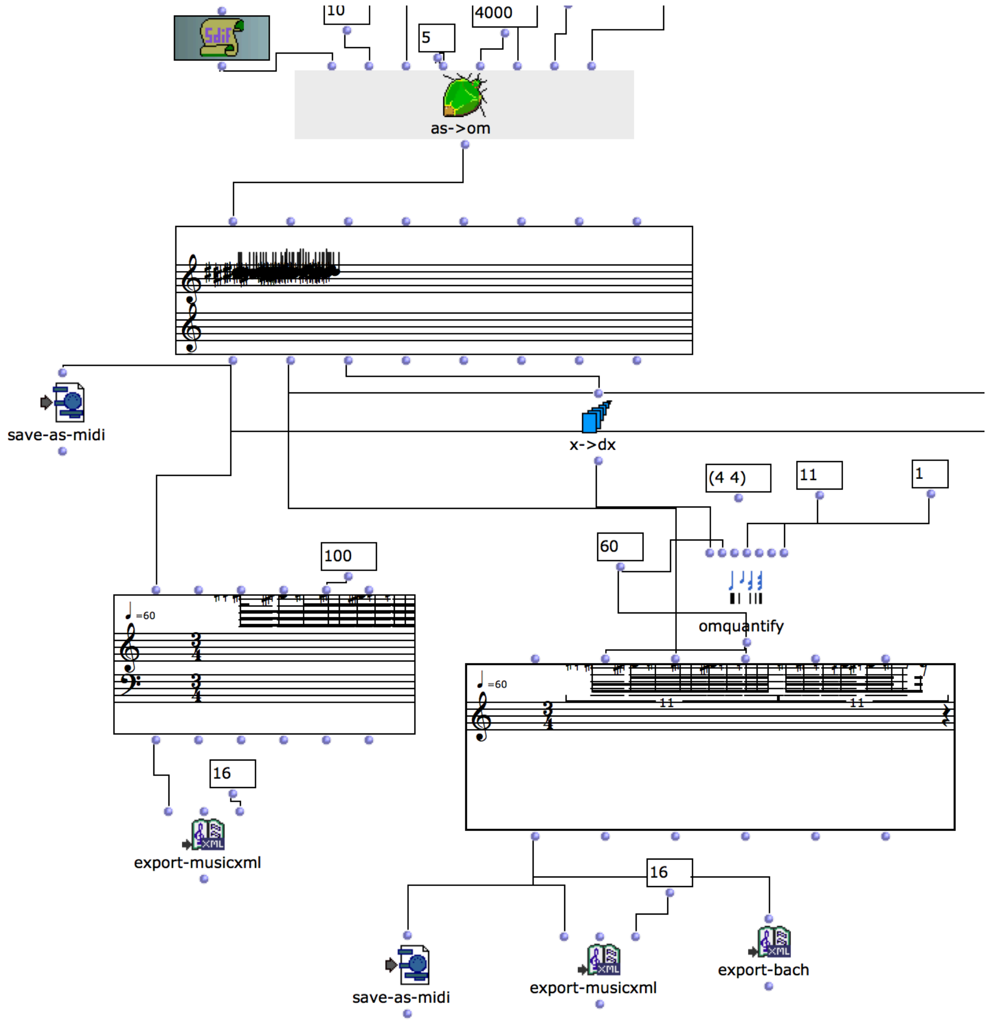


Figure 125. OpenMusic partial tracking transcription patch using the "voice" object (Reid, n.d.)

4.2.2.3.2. Transcription and Notation

The entirety of the piece derives from manual-aural transcriptions of the selected sound models in the field recordings, consisting of cricket and katydid stridulating melodic patterns. The transcription process for this piece consisted of the following steps:

1. Generation of spectral sonogram analyses in AudioSculpt

A sonogram was derived for each of the five microphone field recordings of the summer dusk. The settings were the following:

a) The lowest difference between the sinusoidal components of the crickets' stridulations is ~6 Hz.
b) At a 44,100 Hz sampling rate, the lowest frequency difference of 6 Hz corresponds to approximately 7350 samples.
c) A window size of 36,750 samples (5 times 7350 samples) separates the adjacent sinusoids of the recording.

2. Manual-aural calculation of the exact frequency to be transcribed, using the software's "tuning fork" analysis tool over the sonogram, and quantizing the frequency to eighths of a tone.

Most of the transcription was focused on the stridulating tunings of the crickets (sound model explained above, in 4.2.2.2.3). As explained in the sound model section, the melodic contours of the sound models are characterized by a constant transformation and superimposition of microtonal modes around Bb6 (and its microtonal neighbors below and above Bb6). The quantization of the midi notes, therefore, was adjusted based on the sound models' microtonal resolution as well as the microtonal capabilities of the reed instruments for which the piece is written, which is 1/8 of a tone. The following example (Figure 126) is the microtonal key of symbols used in the manual-aural transcription and presented in the score for *Night Music*. It is a compromise between the detail needed in the descriptive dimension of the score and the instrumental affordability needed in the prescriptive dimension of the score.

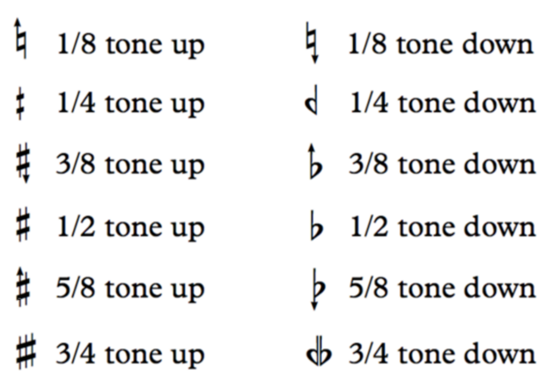


Figure 126. *Night Music,* microtonal key

3. Musical notation transcription of the pitch content in a spatially distributed score

The transcription process for the multitude of pitches that together constitute the cricket chorus began as a hybrid between determinate and proportional notation. Approximate rhythmic values ranging from 32nd notes to half notes were placed over a proportional time grid in the transcription manuscript. Instead of using stemless note heads or uniform stemmed note heads consistent with proportional notation, rhythmic values were used in order to save time beaming note heads to indicate their duration. The rhythmic values placed on the proportional grid effectively provided a visual representation of time placement and duration of the sound models in the manuscript (Figure 127a).

In the subsequent stages of work for this piece (composition and engraving of the score), the symbology was standardized according to proportional notation principles. Mainly, the rhythmic values used in the transcription manuscript (Figure 127a) were replaced with beams of varying sizes representing durations (Figure 127b).

In the following figures, a comparison between the manuscript and the score in the beginning of the 2nd movement is provided. Note the differences between the rhythmic values used in the manuscript for the aforementioned practicalities and the resulting proportional notation of the final version of the score.

Figure 127a. *Night Music,* manuscript, mvmt. 2.
Transcription from the sonogram analysis to proportional/determinate notation (2nd movement).
Note: each "rig" written above each instrument refers to the microphones from which the transcription is derived

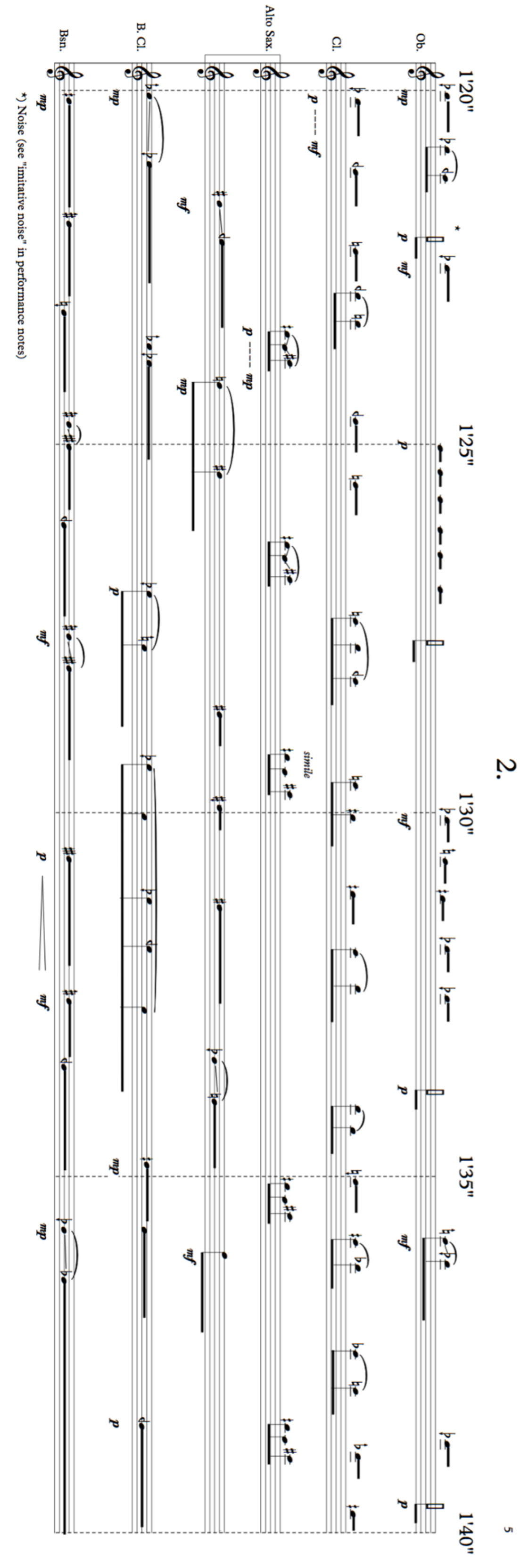


Figure 127b. *Night Music*, score, mvmt. 2, final proportional notation

4. Indeterminacy and Improvisation

There are two points of *Night Music* where the performers are required to go a step further in the embodiment of the biophonic sound models of the cricket chorus and the katydid pulses. The degree of expertise and improvisational talent of each of the members of Splinter Reeds (who premiered the piece) allowed the exploration of how the notated material and the source soundscapes in the field recordings were internalized by the performers. The ultimate goal of the entirety of this dissertation is the composers', performers', and audiences' embodiment –perhaps incarnation– of nature through its sonic patterns. From a performer perspective, internalizing the musical information of the summer dusk biophony opens the possibility of channeling that sound world spontaneously. This spontaneous channeling of the natural sound world may mean different performance strategies depending on the performers' skill sets, from playing without the score and taking musical licenses that honor the sound world of the sound models, to improvisation based on the score's sound world. The latter was the strategy adopted in the score for Splinter Reeds.

Movement 3 features the most melodic section of the piece, in which the individual stridulations of the crickets combine in a magnificent polyphony of phrases in constantly changing modes with a tonic on Bb6, as mentioned in 4.2.2.2.3. After 1'20" of playing this movement, the instruction for the performers for the next 45" is: "Preserving the pitches, dynamics and durations of the preceding materials in this movement, improvise by altering the order of events." The last portion of the instruction ("…by altering the order of the events") was written for subsequent performances in which the performer is not experienced in improvisation. In rehearsal with Splinter Reeds, that last portion was omitted. In the performance and recording of *Night Music*, there is a seamless flow from the notated material derived from the cricket and katydid transcriptions to Splinter Reeds' improvisation based on the sound models.

Movement 5, the most saturated section of the piece, in which the loud katydid pulses dominate the soundscape in a cacophony that surrounds the listener in the forest, the notation is an algorithmic abstraction of the sonic cacophony. Instead of writing the detailed time location of each set of pulsed noises that each of the five instrumentalists are performing, which is the case in movement 4 (where the pulsed noises are less frequent and still do not create a cacophony), the notation in this movement consists of variable rhythmic cells contained between repeat signs for a given time segment (Figure 128, below). The instructions are the following:

1) The upward/downward facing arrow applies to both the noised rectangular note-heads (see explanation in the following page) and the round note-heads presented as a chord. For noised rectangular note- heads, constantly vary the register; for the defined pitches, choose one of them on every repetition.
2) Imitative noise rhythmic choices. Choose between the two rhythms at the given approximate tempo. After playing the chosen motive, immediately or after a brief pause for breathing move on to the pitched material.
3) Pitched material duration. Play a single chosen pitch with any of the given durations. After playing the chosen pitch, repeat the indeterminate material cell until the next time cue.

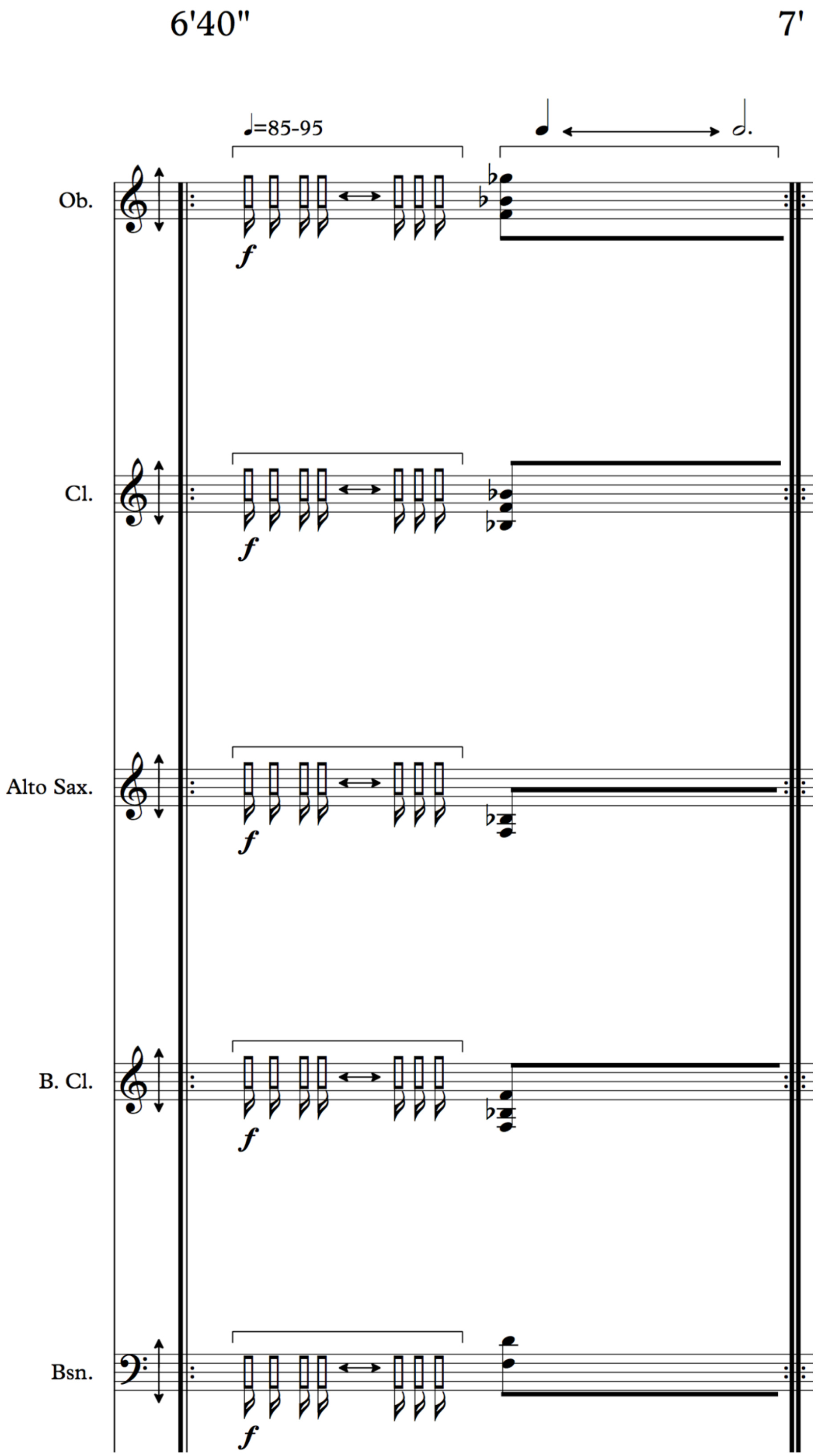


Figure 128. *Night Music,* Movement 5, indeterminate material cells

4.2.3. Under the Sea Ice: motivic and contrapuntal analyses and translation of songs of Arctic bearded seals

Under the Sea Ice is scored for string quartet and electronics. The material for the string quartet derives from transcriptions of six types of bearded seal calls (Figure 129, below) recorded between September and June over three years (2006–09) along the continental slope break in the Chukchi Sea, 120 km north-northwest of Barrow, Alaska. The electronics consist of sea ice recordings in the seals' environment. The recordings were made by Joshua Jones, marine biologist at the Scripps Institution of Oceanography, who provided the audios as well as his article "Ringed, Bearded, and Ribbon Seal Vocalizations North of Barrow, Alaska: Seasonal Presence and Relationship with Sea Ice" (Jones et al., 2014) for the composition of this piece. The article presents qualitative and quantitative information for each of the six types of bearded seal calls, such as seasonal occurrence, types of vocalizations (i.e., breeding and territoriality, mainly), and categories (trills, moans, ascents and sweeps).

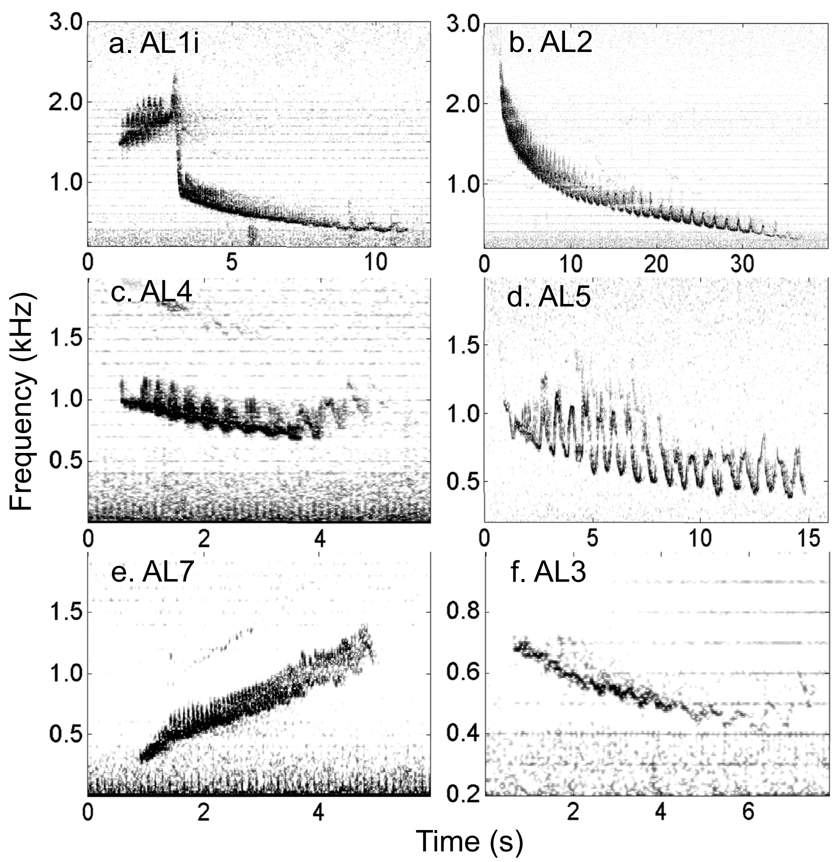


Figure 129. Bearded seal call spectrograms representing the major call types found in Chukchi Sea 2008–09 recordings. Sampling rate: 10.67 kHz, FFT: 1000, overlap: 80%. (Jones et al., 2014, p. 211)

Of the four pieces included as the methodological examples of Environmental Sonic Translation, this is the only piece in which the distribution of events over time and form are not derived from a soundscape of the equivalent duration of the piece. While *Geysir, Waves Break Aural Shores*, and *Night Music* were all formally defined by the development of the events of their source environmental soundscape over time, *Under the Sea Ice* is composed with isolated recordings of the calls that function as sole sound models for the piece. This is due to my interest in understanding the complex and virtuosic intervallic contents of each bearded seal call and building the piece from the particularities—of each call—to the generalities—the bearded seal cacophonic chorus.

The transcriptions of the six sound models presented such organicism and motivic similarities that it became clear that the piece could be constructed with counterpoint techniques "emphasizing the melodic or 'horizontal' as opposed to the harmonic or 'vertical' domain" (Rushton, 2001), obviously not based on functional harmony and the rules of consonances and dissonances not applicable to natural sound. Augmentation, imitation, transposition and textural development are the main techniques applied to the bearded seal sound models.

4.2.3.1. Sound Models

I manually-aurally transcribed the six bearded seal call sound models using SPEAR (see FFT settings in the General Analytic Procedures sections "Sonogram Analysis", 4.2.1.1., and "Manual-aural transcription in SPEAR", 4.2.1.3.) Based on Jones's classification of calls, the sound models consist of sweeps (descending), ascents, and trills. Proportional notation is used in the sound model music notations provided in this section due to aesthetic and practical considerations (see "Proportionally Notated Transcription", 4.2.1.5). In all six seal calls, each system is five seconds in duration, with five 1 second-long bars, displaying the time cue (as a number indicating the seconds) at the beginning of each system. Stems do not indicate rhythmic values and are used as visual aid in the context of the proportional notation. Microtonal glissandi are a constant and distinctive feature in the bearded seal songs. The six transcriptions shown below are included in the parts for the string quartet together with the bearded seal song audios in order for the performers to study each of the calls individually and gain the deepest possible understanding of the songs.

4.2.3.1.1. *AL1i2, AL1i-variation, and AL2*

The most salient feature in these two calls is a long descending sweep varying between 35 and 50 seconds. While AL1i2 typically begins with a brief ~5" ascent before the long descending sweep, AL2 begins the sweep at its highest frequency.

A unique feature of these two bearded seal songs is the gradually expanding time and frequency intervals of ascending/descending glissandi nested within the main overarching descending glissando line. In the examples below (Figure 130a and b), notice the increasing intervals (both in time and frequency from the end of the 1st third to the end of the sonograms.

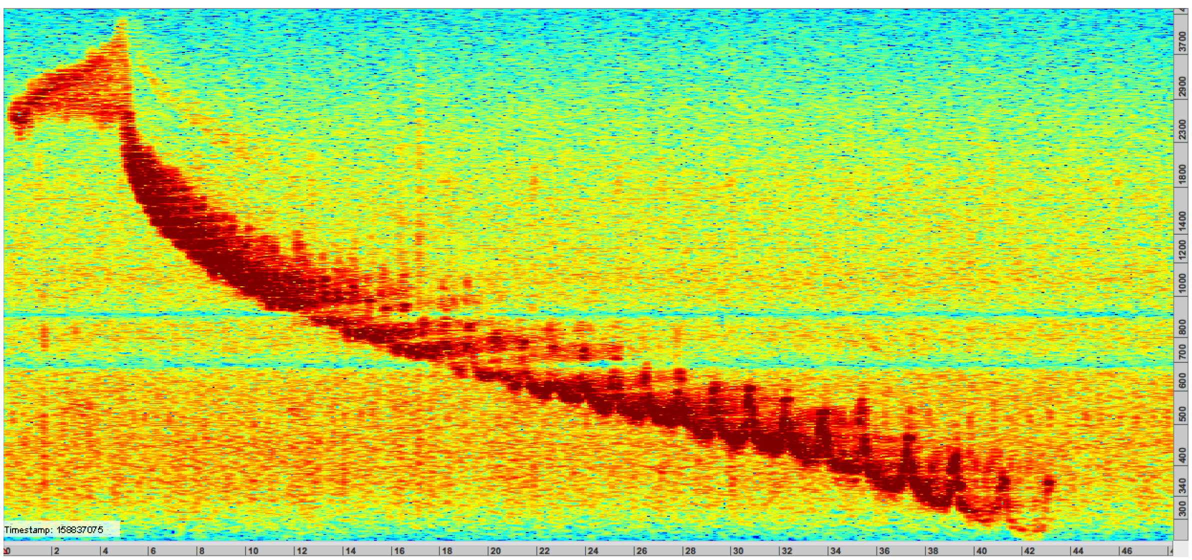


Figure 130a. Bearded seal call "AL1i2", gradually expanding glissandi nested within the main descending glissando (logarithmic frequency display)

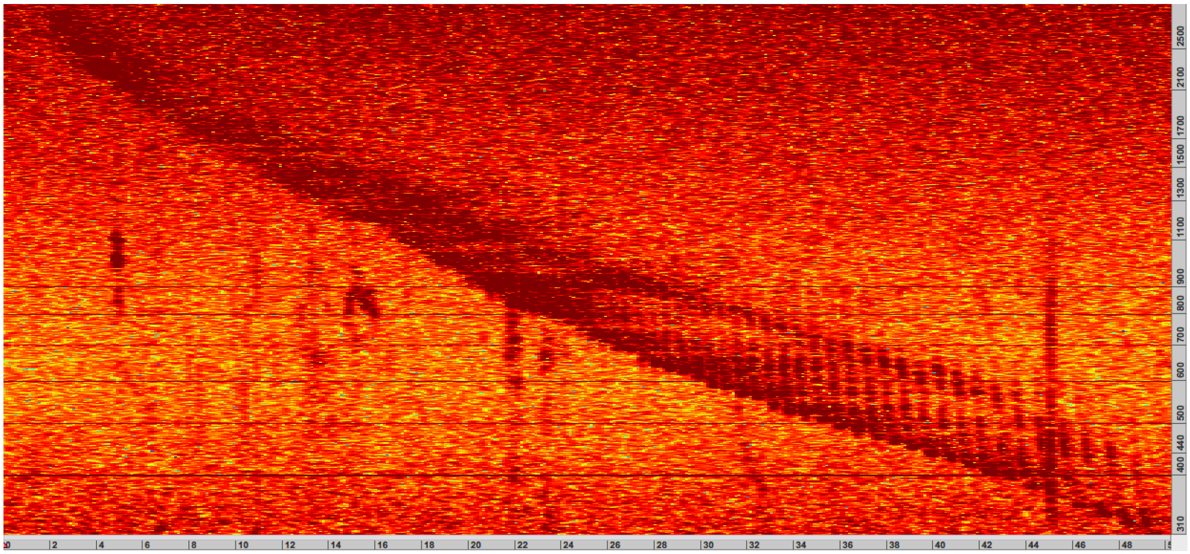


Figure 130b. Bearded seal call "AL2", gradually expanding glissandi nested within the main descending glissando (logarithmic frequency display)

In the transcriptions below, there is a gradual intervallic increase. In AL1i2, the increase is more complexly organized than in AL2, in which the linearity in the flow from microtonal seconds to fifths can be perceived almost at first sight of the notation.

The main difference between the AL1i2 and AL2 types of calls is that AL1i2 includes a combination of the descending sweep and an ascent, while AL2 consists entirely of a descending sweep. The two versions of the AL1i2 call differ in the placement of the ascent. The first version (recorded by Jones) begins the song with the ascent followed by the descending sweep. The second version begins with the descending sweep, followed by the ascent in the middle, and ending with another sweep.

Since an early stage of the pre-compositional work of the piece, the contrapuntal potential of these temporally and micro-intervallically expanding glissandi suggested that the climax of the piece would consist of a highly saturated polyphony with superimpositions of mostly "AL1i2", "AL1i-variation", and "AL2" calls in different transpositions.

AL1i2

Figure 131. Bearded seal call “AL1i”, transcription

AL1i–variation

This is the only call that was not recorded by Joshua Jones. It was made by Ray, Watkins and Burns, published in *The underwater song of Erignathus*, published in Zoologica journal in 1969 (Ray et al., 1969).

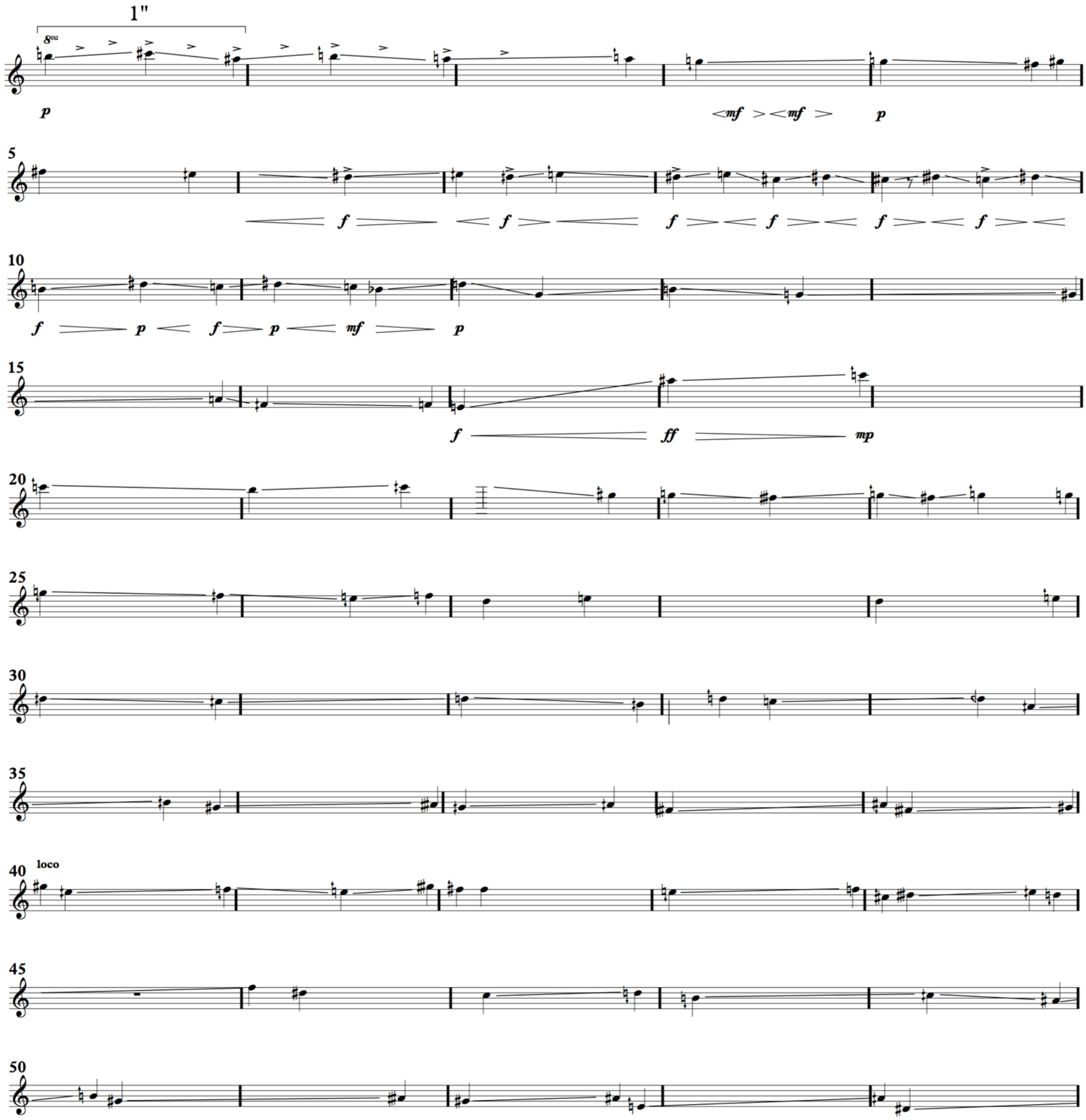


Figure 132. Bearded seal call "AL1i–variation_Ray, Watkins and Burns", transcription

AL2

Figure 133. Bearded seal call “AL2”, transcription

4.2.3.1.2. *AL7*

AL7 is a short call that consists of a two-octave ascent in 5 seconds. It is the second most commonly used sound model in the piece, after AL1i and AL2.

Figure 134. Bearded seal call "AL7", transcription

4.2.3.1.3. *AL4 and AL5*

These calls, consisting mainly of trills and between short and moderate duration in relation with the AL1i and AL2 calls, were used only once in the piece.

AL4

Figure 135. Bearded seal call "AL4", transcription

AL5

Figure 136. Bearded seal call "AL5", transcription

4.2.3.2. Manipulations of the sound models

From a compositional perspective, the approach to the bearded seal sound models has a higher degree of agency and manipulations than the approach in *Night Music*, which leaves the transcriptions of the sound models without significant alterations. The intervallic richness and timbral complexity of the bearded seal calls, especially AL1i2 and AL2—the most prominently explored in the piece—led to manipulations that had a twofold intention: (1) to explore the materials themselves through different time and frequency dimensions such as time-stretching and transposition, and (2) to develop the material as a narrative strategy within the overall form. Such manipulations are covered in this section. Explorations were first performed electronically and then transcribed to music notation.

When applicable, nomenclatures for the following manipulations consist of two processes that are equivalent in signal processing and in music notation. For example, time stretching is in signal processing what augmentation is in counterpoint notation.

4.2.3.2.1. Time stretching / Augmentation

The timbral complexity perceivable upon listening and viewing the sonograms and re-syntheses of the ascents in models AL1i2 and AL7 led to explorations of the ascent through time-stretching of the re-synthesized sinusoids. The main motivation was to reveal the harmonic intervallic relationships between the closely adjacent sinusoids that create the unique timbral roughness characteristic of the ascents in the models. In the examples below (137a–c), the re-synthesized bearded seal call "AL1i2" is manipulated in the following order of processes: (1) Re-synthesis, displaying only partials over -30 dB; (2) Time-stretch four times the original duration; (3) Transposition to an octave lower. This last process of transposition was applied for instrumentation purposes, considering a lower register in the string quartet that would be richer in harmonics. Richness in harmonics was pursued in order to emulate the timbral complexity of the bearded seal sound models.

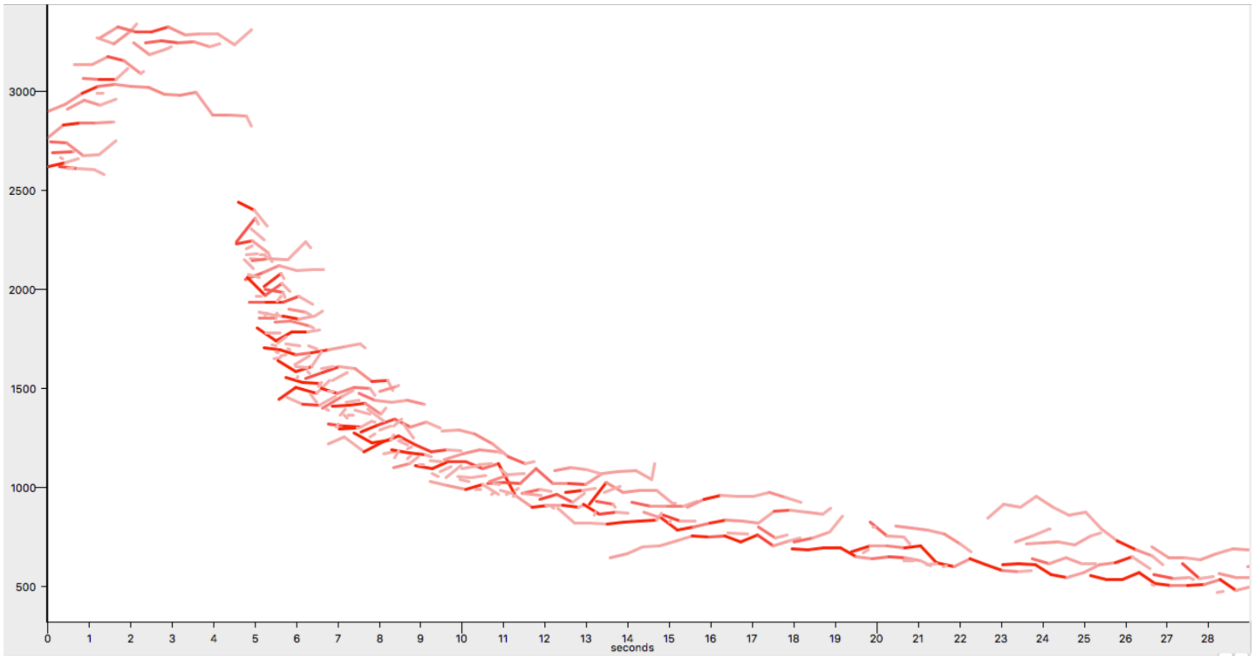


Figure 137a. Bearded seal call “AL1i2”, Re-synthesis displaying only partials over -30 dB

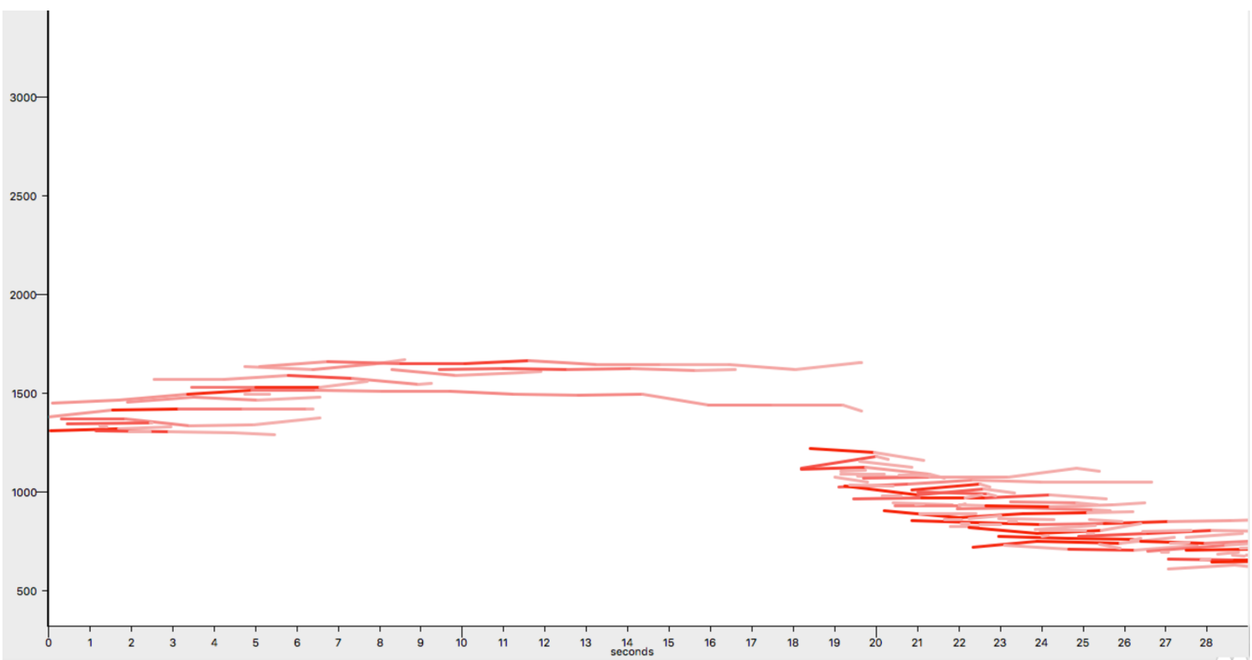


Figure 137b. Bearded seal call “AL1i2”, Time-stretched four times the original duration

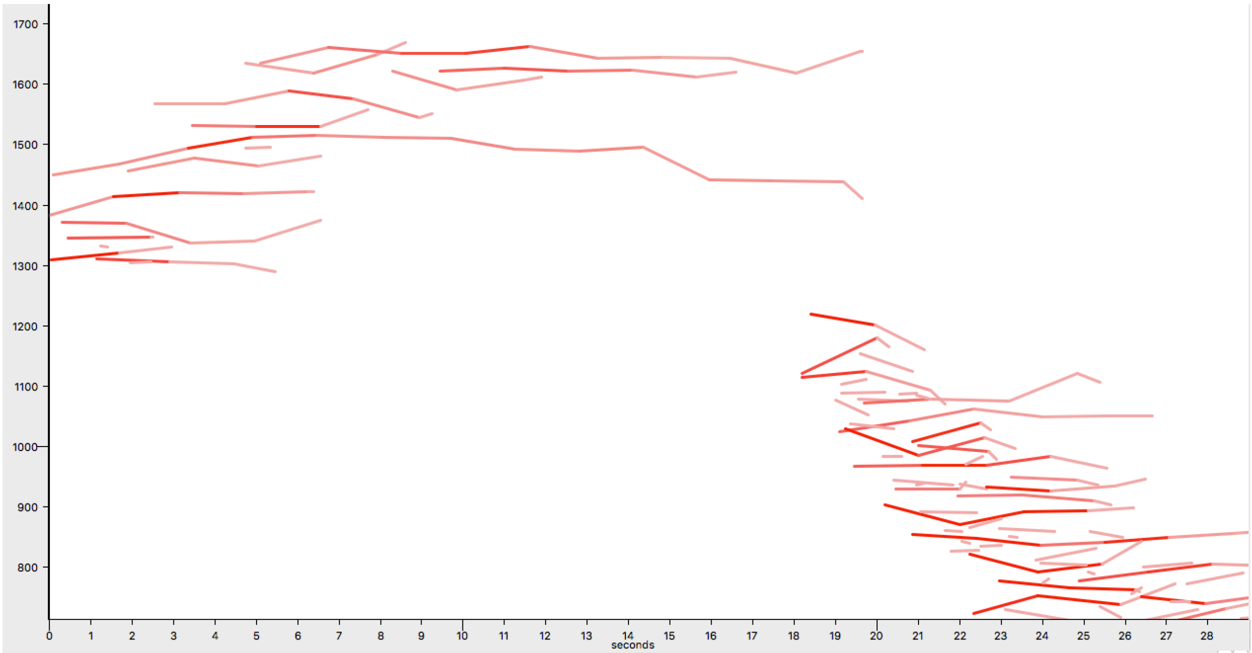


Figure 137c. Bearded seal call “AL1i2”, transposed an octave lower

The re-synthesized, time-stretched, and transposed audio of the seal call was then transcribed into proportional/spatial notation for the string quartet, with a pitch resolution of eighths of a tone. The continuous sinusoidal motion within the overall ascending line over ~20 seconds was represented with an indeterminate notation for Violin II, Viola and Cello consisting of irregularly shifting the given notated pitch around a quarter of a tone throughout the duration of the bar.

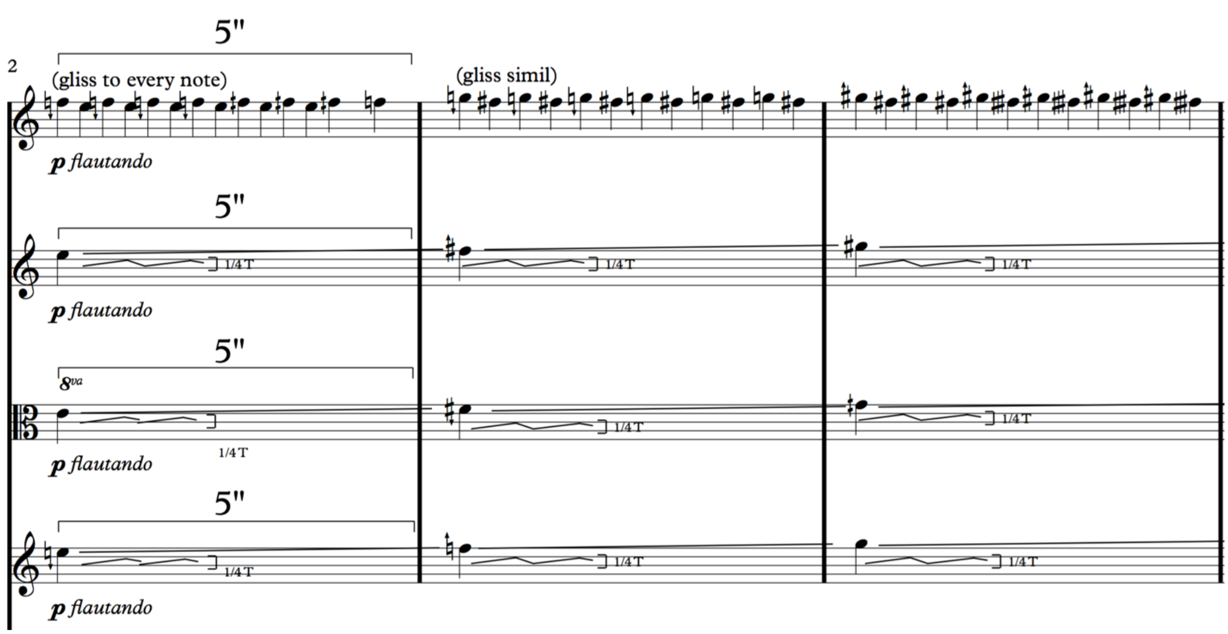


Figure 138. Bearded seal call “AL1i2”, transcription for the string quartet (00:00–00:15)

4.2.3.2.2. Pitch shifting / Transposition

There were mainly two processes for pitch shifting the bearded seal calls: (1) re-synthesized calls-based pitch shifting in SPEAR, which was exclusively applied to the time-stretched pitch shifted calls (see 4.2.3.2.1.), and (2) re-synthesized calls-based pitch shifting in Logic, applied to all non-time stretched calls. The latter process was applied mainly in the 4th section of the piece, characterized by a polyphonic texture that is gradually increasing in density. The pitch shifting in this section had the purpose of exploring different registers of the bearded seal calls and the strings in counterpoint. A pitch shifter was added to each Logic track assigned to each instrument. Manual automation was applied, specifying the pitch shifting by number of semitones. In the example below (Figure 139), semitone numbers above and below "0" show the pitch-shifted automation applied to each track. The regions in each track in the example consist of different seal call types as well as portions of them such as ascents (AL2, AL1i, AL1i2, AL1i time-stretched ascents).

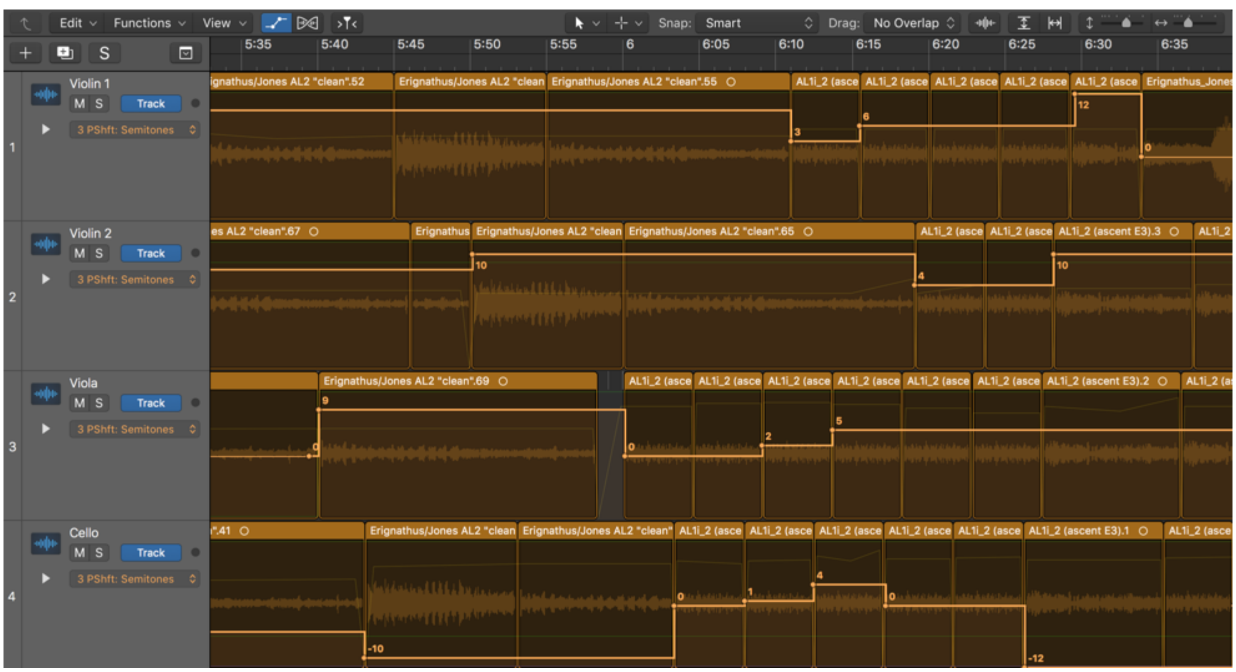


Figure 139. *Under the Sea Ice*, maquette in Logic, pitch shifted bearded seal calls

After pitch shifting the sound models, transpositions were made to the original transcriptions of the seal calls (see 4.2.3.1.) and placed in the same order of the Logic bearded seal call maquette.

In Figure 140a, below, the beginning ascent in the original AL1i2 model is shown in the first 4 seconds of the call (as mentioned above, each bar represents a second in duration). The transpositions in different registers of the string quartet are shown in Figure 140b, in which the beginning ascent of AL1i2 is played by each instrument at asymmetrical points in time. This procedure of transposition in pitch and time is common throughout the piece, especially in the polyphonic sections of the piece.

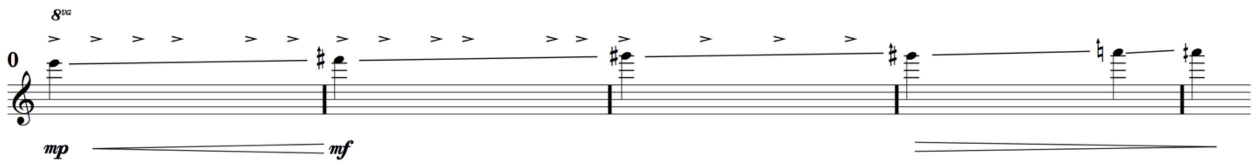


Figure 140a. AL1i2 sound model, transcription of the beginning ascent (seconds 0 through 4)

Figure 140b.
Under the Sea Ice, transpositions of the beginning ascent of AL1i2 sound model, fourth section of the piece

4.2.3.2.3. Motivic fragmentation

In addition to the objective of exploring the bearded seal calls contrapuntally, the polyphonic section of the piece sought to recreate the underwater environmental soundscape of the bearded seals. While earlier sections of the piece consists of experimentation around individual bearded seal call sound models, this section is a recreation of the cacophony resulting from the hundreds of seal calls occurring simultaneously under the sea ice. For this purpose, the frequency of the bearded seal call onsets was increased and portions of the seal calls with larger intervals were made more frequent. From an instrumentation perspective, the six bearded seal call sound models with some transpositions would have sufficed to create the density needed in this section if the piece had been written for a string orchestra. Each of the transpositions would have been assigned either individually or in small groups in *divisi*. A very small latency between onsets could have been achieved due to the number of performers available. In the string quartet, however, in order to have small latency between onsets, the calls needed to be fragmented, interrupted, in order to create the illusion of a larger diversity and simultaneity of calls, with each performer beginning a new call more frequently than what the original seal call durations allow.

In Figure 141a, below, the entire AL2 sound model transcription is shown, with the squared area of the call showing the portion of the material that is used as the fragmented material during the polyphonic section of the piece. Figure 141b shows the way in which the fragment of AL2 is inserted in the score for the viola and cello parts.

Figure 141a. *Under the Sea Ice*, AL2 seal call, and selected fragment

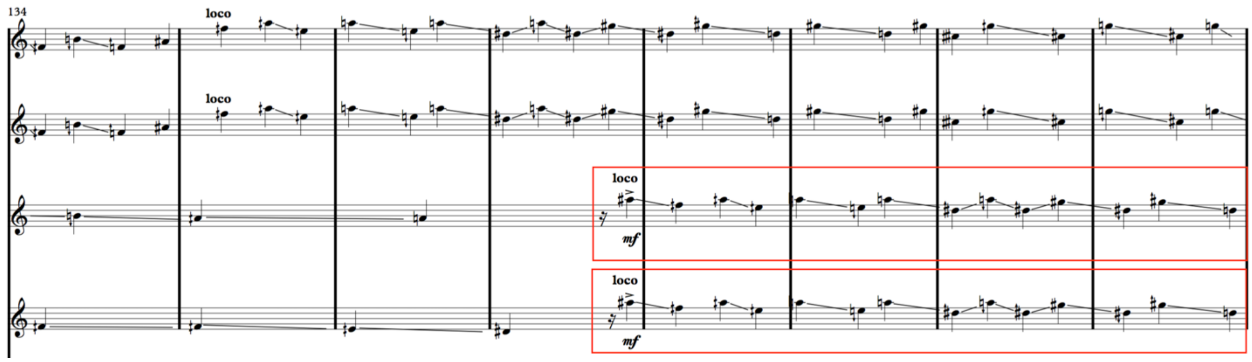


Figure 141b. *Under the Sea Ice*, AL2 fragment insertion in the score (1 bar = 1 second)

4.2.3.2.4. Proportional notation-derived texture and timbre

In sections of the piece in which all the parts are playing the same material, the beat-based (60 BPM) proportional notation results in a loose synchronization that functions as heterophony. The slight asynchronies between the parts results in micro-clustered vertical relationships that seek the textural, timbral and harmonic complexity in the bearded seal call sound models. In the example below (Figure 142), two slightly varied versions of AL2 are played simultaneously by all the parts in the string quartet. The upper two parts are the original model, while the lower parts are a simplified version of the model –without the micro-intervallic glissandi nested within the main overarching descending glissando line. The sonic result of the impossibility of precise synchronization of these materials is a micro-clustered vertical harmony aligned with the timbre of the seal calls.

Figure 142. *Under the Sea Ice*, bar 118-121, AL2 *tutti* (1 bar = 1 second)

Another material of the piece in which the complex timbre is achieved by assigning the same material to three or four parts is the AL1i-variation ascent fragment (Figure 143).

Figure 143. *Under the Sea Ice*, bar 37-39, AL1i, fragment–ascent (1 bar = 1 second)

In the example below (Figure 144), the final section of the piece consists of AL2 played *tutti*. Unlike the previous example, in which there is a slight variation in the notation, this section does not feature a variation in the notation. The relatively rapid moving glissandi (approximately four per second) performed by each instrument in slight a-synchrony generates the complex texture heard in some of the underwater soundscapes of the bearded seals' environment.

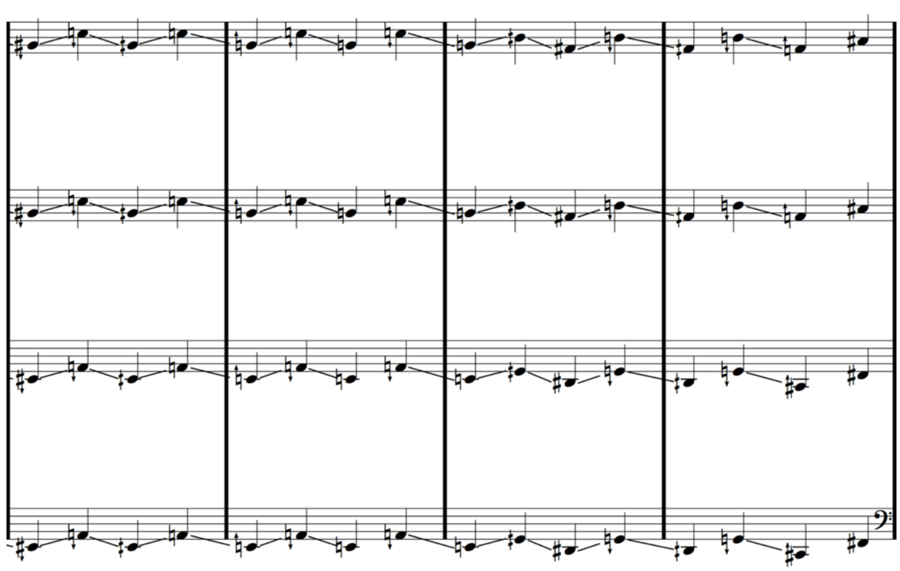


Figure 144. *Under the Sea Ice*, bar 396-399, AL2 *tutti* (1 bar = 1 second)

4.2.3.3. Form

The form of *Under the Sea Ice* is the result of a detailed exploration of the bearded seal sound models (4.2.3.1.) and the materials derived from their manipulations (4.2.3.2.), both individually and in combinations. Different sections of the piece focus on specific areas of examination of each sound model, including time-stretching, pitch shifting, micro-intervallic harmonic production through unison passages, and polyphonic arrangements. The general formal curve of the piece is the flow from clear and focused presentations of the isolated seal calls to their fragmentation and superimposition in a complex polyphony that aims to re-construct the complexity of the soundscape under the sea ice.

The example below (Figure 145) is the maquette for the piece with the formal descriptors by sections, call types, and manipulations at the top. Containing all the re-synthesized bearded seal calls, it stands alone as a *musique concrète* collage of the sound models and their manipulations. At the same time, it served as a sounding board for harmonic, density, and formal relations.

Section	A	B	C	D	C'	E	C''	D'
Call type	AL1i2	AL1i	AL7	AL2	AL1i2	Polyphony	AL7	AL2
Manipulation	stretched	none	stretched	none	transp.	combined	comb.	none

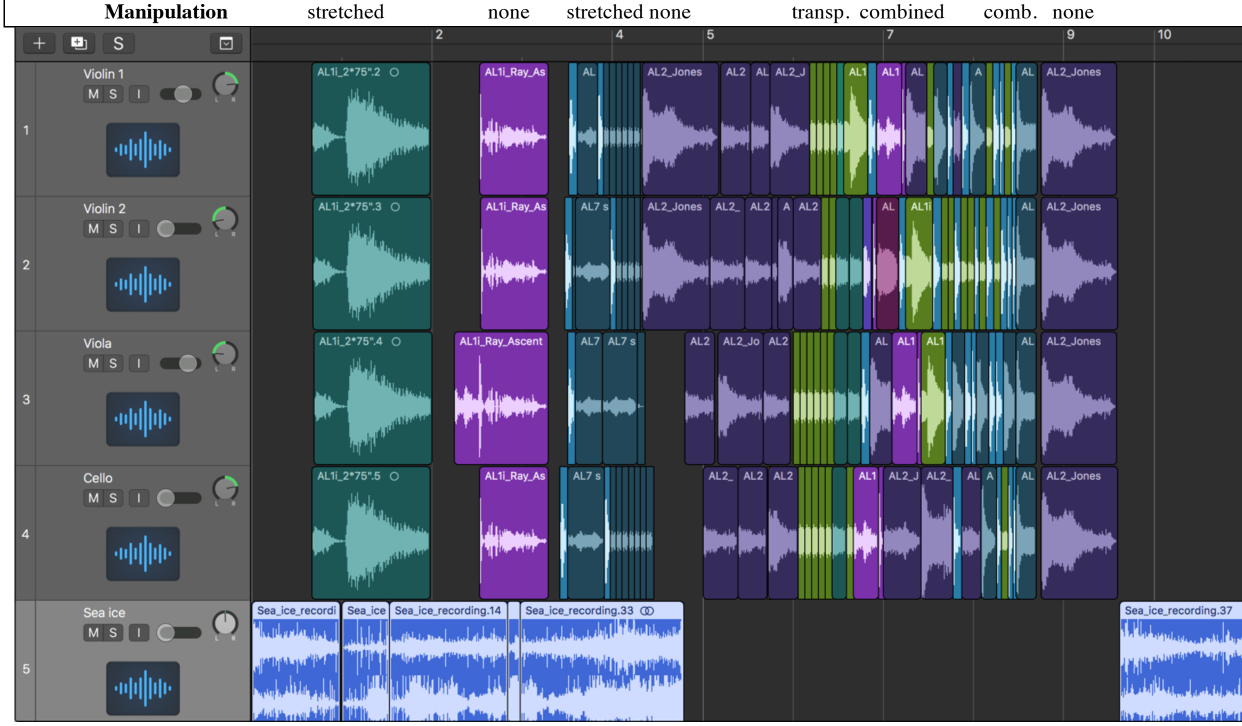


Figure 145. *Under the Sea Ice*, maquette. Each sound model is color-coded and labeled (AL1i2, AL1i, AL7, etc.)

The following figures show a more detailed display of the maquette.

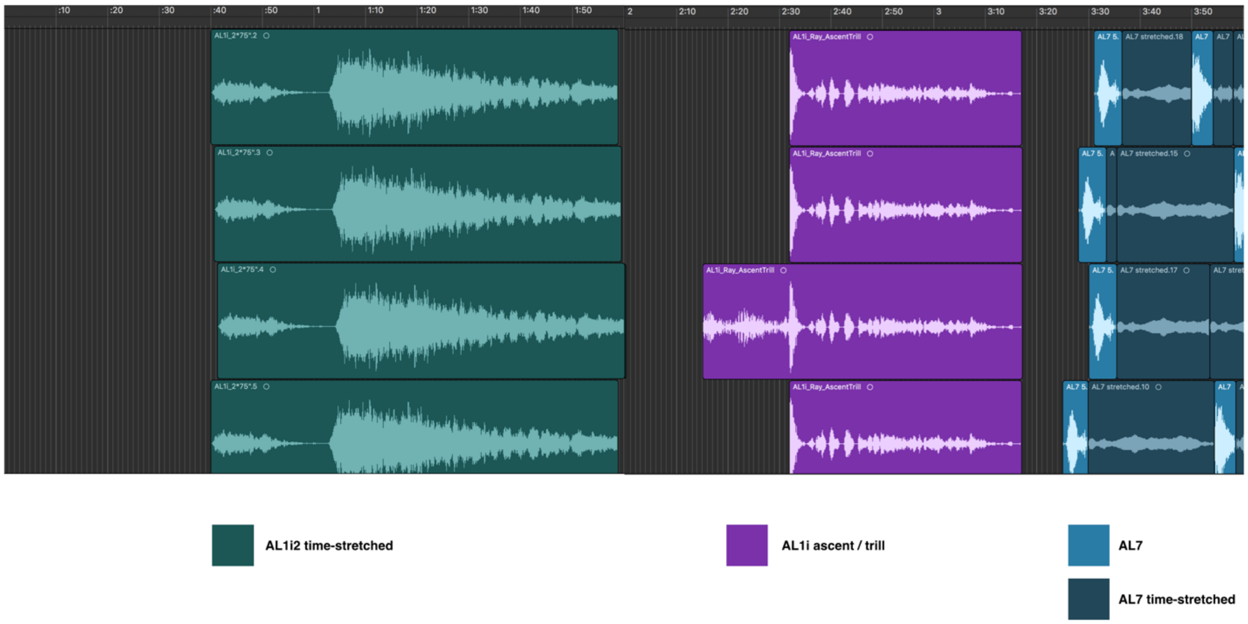


Figure 146. *Under the Sea Ice*, maquette, sections A, B, and C (fragment)

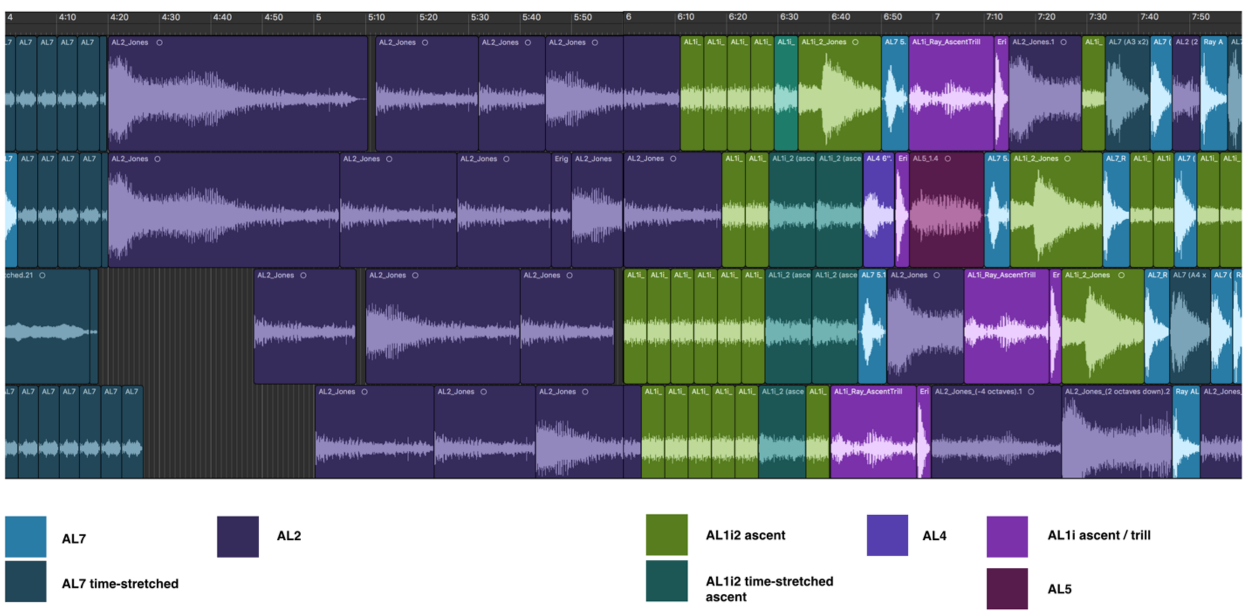


Figure 147. *Under the Sea Ice*, maquette, sections C (fragment), D, C', E (fragment)

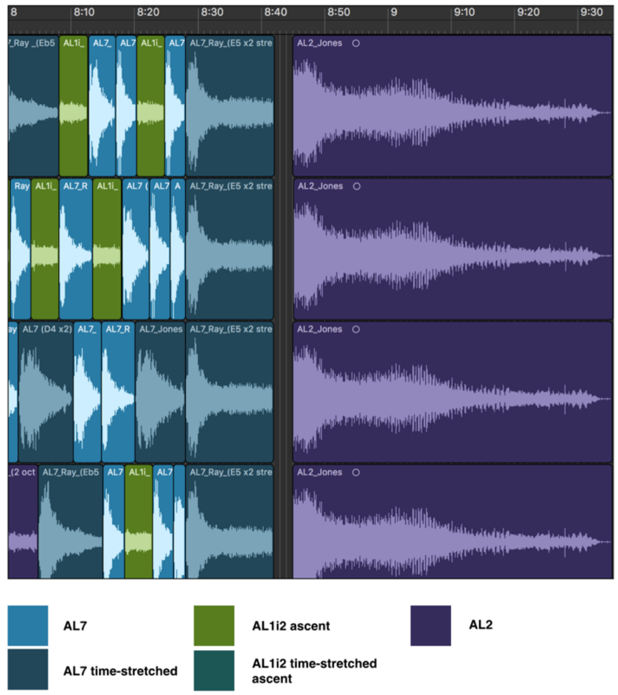


Figure 148. *Under the Sea Ice*, maquette, sections E (fragment), C'' and D'

The sections below explain the main compositional ideas in each section. The musically transcribed sound models correspond to each of the color-coded regions displayed in the maquette.

4.2.3.3.1. Section A: "AL1i-2", time-stretched (augmentation)

The opening section features the call "AL1i" time-stretched 2.75 times its original duration, in a microtonal heterophony that results from indeterminacies in some notated parameters of a passage in unison. The indeterminacies consist of (1) the relative temporal flexibility that the performers have for playing the events between time cues (each 5" bar); (2) the durations and intervals of the microtonal glissandi, indeterminately notated as] 1/4T.

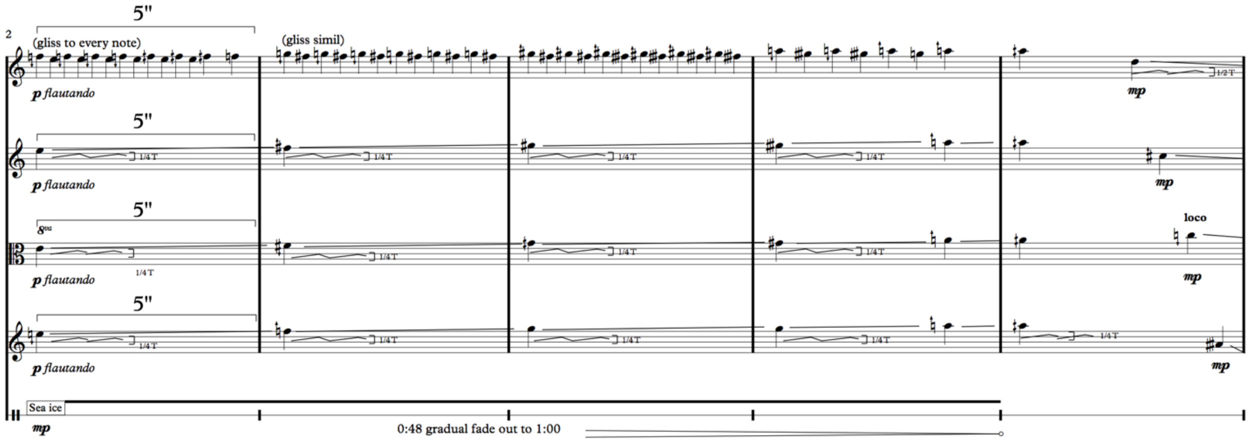


Figure 149. *Under the Sea Ice*, Section A

4.2.3.3.2. Section B: "AL1i-variation"

This is the first time in the piece in which a bearded seal sound model is presented entirely and without manipulation. The idea in this section is to play with instrumental texture, from solo to tutti, and explore the micro-intervallic harmonies resulting from a single material performed simultaneously in proportional notation (as explained in 4.2.3.2.4.). The first segment of the model is introduced by the Vla. solo. At the beginning of the 2/3 of the piece, Vl. I and Vl. II join in the glissando, in unison. The Vc. incorporates five seconds later, ending the presentation of the AL1i-variation sound model in tutti. The example below (Figure 150) shows the transition from solo to tutti in the section.

Figure 150. *Under the Sea Ice*, Section B, solo to tutti transition

4.2.3.3.3. Section C: "AL7", melody (real time) and accompaniment (time-stretched)

Another approach to exploring micro-intervallic relations consisted on a melody and accompaniment idea. The time-stretched partials of the ending of AL7 are assigned to three instruments, while a solo instrument performs the original sound model without time-stretching.

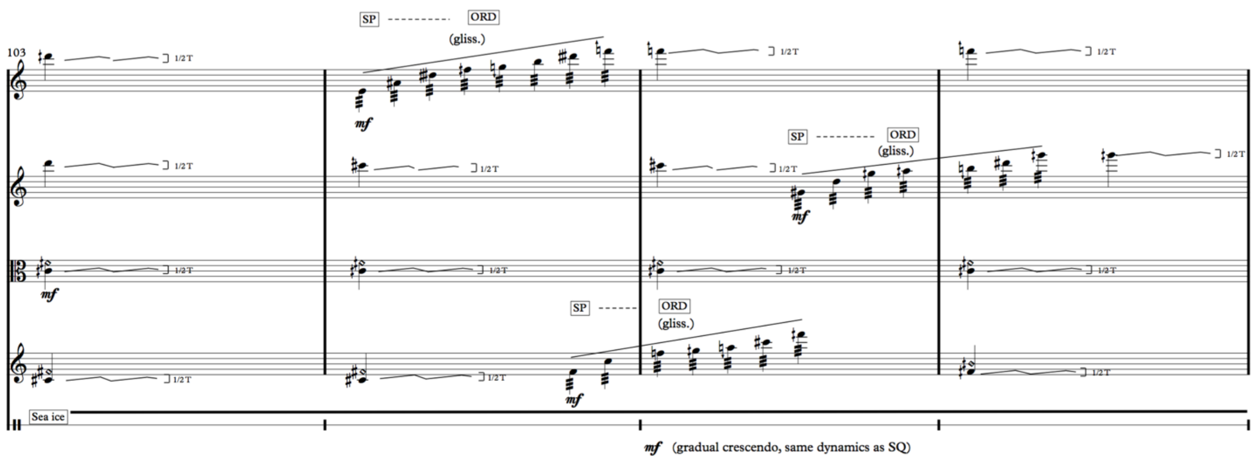


Figure 151. *Under the Sea Ice*, Section C, melody (real time) and accompaniment (time-stretched)

The sea ice field recording consists of the sounds of fissures in the ice, which are timbrally analogous to the highest register partials of the AL7 sound model.

4.2.3.3.4. Section D: "AL2", original, simplified, and fragmented

Similarly to section B, this section presents an entire sound model in real time and explores the timbral and textural results of assigning the same proportionally-notated material to various instruments. As a means to generate a higher degree of micro-intervallic harmonies, a simplified version of the model was superimposed to the original. Similarly to a Schenkerian *urlinie* applied to a sound model[12], the simplification consisted of eliminating the ascending/descending glissandi nested within the main overarching descending glissando line. The resulting material is a single

[12] Schenker conceived the *Urlinie*, the "fundamental line", as a kind of motivic line characterized by its fluency, repeated under different guises throughout the work and ensuring its homogeneity. He later imagined that a musical work should have only one fundamental line, unifying it from beginning to end (Schenker, 1979, p. 10).

descending glissando line. The contrasting superimposed material is shown in the last two measures (or seconds) of the example below (Figure 152).

Figure 152. *Under the Sea Ice*, Section D, original AL2 vs simplified AL2

At approximately the middle of the AL2 sound model, the unison of the quartet splits in two groups, marking the beginning of the polyphonic section of the piece. The material of the Vla. and Vc. play the AL2 sound model in canon, three seconds delayed in relation to Vl. I and Vl. II, starting at second twenty-five of the AL2 sound model (Figure 153). This is the first time in the piece in which a sound model is fragmented, a process that gradually increases in tandem with the polyphonic complexity towards the climax. As section D progresses, various fragments of AL2 are introduced, and the polyphonic texture is increased from two to four voices.

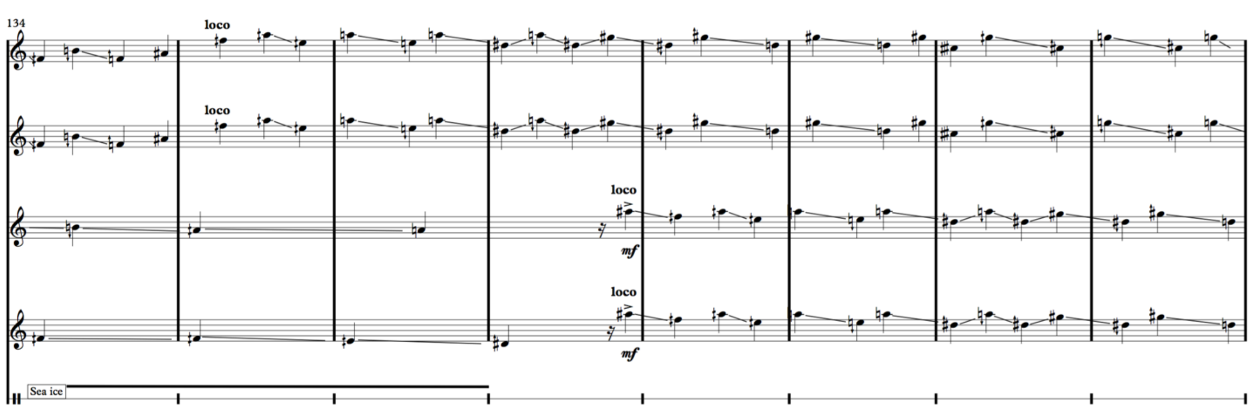


Figure 153. *Under the Sea Ice*, Section D, canon

4.2.3.3.5. Section C': "AL1i2 ascent", transposed

The complex polyphonic texture from section D, directly preceding this section, motivated an expansion in the frequency domain as a means of texture development. The sound models, which up to this point of the score had been presented in their original pitches, are now pitch shifted (transposed) to various registers of the strings. The ascent portion of the AL1i2 model was chosen for contrast purposes, creating a pause in the ascending/descending glissandi patterns before they become a dense cacophonic mass in section E. As mentioned in the *Pitch-shifting/transposition* section of this chapter (4.2.3.2.2.), this technique is used with the intention of creating an illusion of a higher density of sounds through the variety in registers.

Figure 154. *Under the Sea Ice*, Section C', AL1i2 ascent, transposed

4.2.3.3.6. Section E: polyphony of all sound models

All the sound models in multiple transpositions are distributed throughout the full range of the string quartet. This aims to explore the sonic potential of each sound model in different ranges of each instrument, from the first to the fourth string. Section E, together with C'', constitute the climax of the piece, where the maximum degrees of density and dynamics are reached. This section led Joshua Jones (see 4.2.3. for details) to the notion of "a work that conjures the tension between the environment and the animals. The musical translation of the bearded seal songs and the performance of the JACK Quartet imposes for the time an emotional experience that entangles beauty and fear, external pressures with internal responses" (J. Jones, personal communication, February 11, 2016).

Figure 155. *Under the Sea Ice*, Section E, polyphony of all sound models

4.2.3.3.7. Section C'': "AL7" and "AL1i2 ascent", various transpositions

The culmination of the climax, section C'' features the bearded seals' ascents unfolded as polyphonic texture in various transpositions. The momentum accumulated in section E is propagated in section C'', increasing the intensity of the AL7 and AL1i2 ascents. The contrast between the small intervallic shift in AL1i2 (a quarter-tone augmented fourth) and the 2-octave shift in AL7, as well as higher dynamic ranges and a predominance of *sul ponticello* bowings, lead to the climactic resolution of the piece at the end of C'' and beginning of D'. This resolution is characterized by the tutti ***f*** that starts the AL2 call, the last of the piece.

Figure 156. *Under the Sea Ice*, Section C'', AL7 and AL1i2 ascents in various transpositions

4.2.3.3.8. Section D': "AL2", unison-derived heterophony

The last section of the piece is a return to a texturally more transparent presentation of a sound model, returning to an apparent monophony that in reality works as heterophony due to the nature of the notation, as explained in 4.2.3.2.4. The energy of the polyphonic section is preserved by the nature of the sound model, with its inner motion of continuous ascending/descending glissandi within the overarching descent that characterizes it from beginning to end. The proportional notation of the model in *tutti* produces a timbral complexity that most closely translates the original sound model: the vertical asynchronies between the parts playing the same material generates micro-tonal clusters similar to those in the bearded seal AL2 call, clearly depicted in the spectral analyses shown in 4.2.3.1.1. By the end of the sound model, natural harmonics are gradually introduced for each string, creating a sudden deviation from the sonic world of the bearded seals into the sonic world of human instruments and harmony. The vertical relations gradually shift from complex dissonance to overall consonance resulting from the strings' harmonics.

Figure 157. *Under the Sea Ice*, Section D', AL7 and AL1i2 ascents in various transpositions

Final Thoughts and Future Directions

After repeated listenings and reflections of my work, it becomes evident to me that my primary focus has been centered around the extraction of sonic material that I can spiritually commit to develop as composition and orchestration of instrumental music. Natural sources of sound, including those that have been modified or used by humans as tools, might be the only sort of soil that I have found consistently fertile for my notated music composition. In the context of the conclusions to this work, it is important to highlight the analogies of the last two sentences. An important distinction can be extracted from this analogy: the environmental sonic translation concepts and methodologies that I propose are not seeking environmental sonic mimesis; they are deriving quantitatively measured sonic parameters from the environment and using them as compositional material for instrumental music.

Continuing the past fifty years' explorations and concepts of timbre through the fast fourier transform, phonorealism, sound models, and acoustic ecology, this approach to environmental sonic translation joins the hundreds, if not thousands, of musical practices that incorporate natural sounds since ancient times. It does so with the technologies, problems, aesthetics, and challenges of its period—the decades of the 2010s and 2020s. Among these, the inescapable environmental awareness in the face of climate change and its impending possible catastrophes charge this compositional approach with purpose. To engage with acoustic, scientific, and social data, to understand it, and to present it to collaborators across the fields and to audiences, is the minimum result that this kind of work aspires to. The larger goal is to portray in the listeners—hopefully culturally and socially diverse, and not only a niche in universities—a spirit of embracing, honoring, and loving our world, especially nature. In this spirit, the state of wonder and reverence that natural forms—especially sonic—instill in me is the reason for this work.

Since 2018, my involvement with the Coastal Futures Conservatory and the Environmental Resilience Institute, both at the University of Virginia, have resulted in sonifications of various types of data related to climate change. So far, this work has not included the environmental sonic translation tools presented in this dissertation. Non-acoustic data has been its sole compositional source and field recordings are the only commonality between this sonification work and the environmental sonic translation discussed here. Due to the fact that it requires scientific data to be parsed and interpreted, the interdisciplinary nature of sonification makes this sort of work an important opportunity to expand and continue adopting and contributing valuable tools and forms from and to other fields. For this reason, an immediate future direction for environmental sonic translation is its adherence to sonification. This fusion of sonification with environmental sonic translation entails a more complex endeavor due to the processes that would be added to the already large number of steps in the methodologies presented in Chapter IV. However, if collaboration is possible and there are resources for this confluence of music and science, the methodologies here presented could streamline the work of a team of collaborators including students learning ecoacoustics and acoustic ecology, software developers, and researchers in the fields. Such a collaborative project would be desirable especially for universities and institutions interested in generating synergies between arts, sciences, and humanities.

Lastly, I hope this work may find a total integration between the natural sonic data that it studies and the musicians' inherent and intuitive creative impulses. In other words, the ultimate objective is to become the environmental sound, to think of sounds, to produce sounds that share their fundamental origins with nature, without the need of analysis, sonograms, or microphones. I hope that in the future, through the exploration of the work that I have presented here, environmental sonic forms may be integrated in a way in which they are embodied, played without thinking, sung intuitively, and realized as a connected whole.

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(APA Citation Style)

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Appendix

Scores for dissertation pieces / album including the dissertation pieces

Geysir
Waves Break Aural Shores
Night Music
Under the Sea Ice

Aural Shores
Environment-derived composition, Vol. 1
Edgetone Records
April, 2020

https://edgetonerecords.bandcamp.com/album/aural-shores

Geysir

for seven pianists and electronics

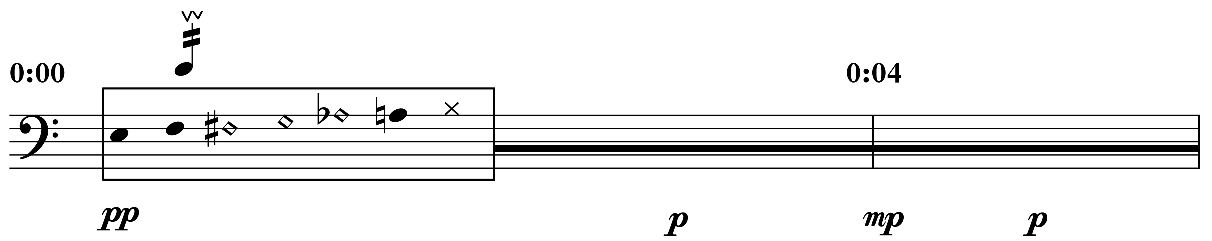
Christopher Luna-Mega (2016/2019)

This piece is a study of the acoustic properties of an Icelandic geyser recorded 180 miles East of Reykjavik in the valley of Haukadalur. Its complex harmonies, dynamics and rhythms shift perpetually and in subtle ways. If listened to as a background, the sound may appear to be static. With a focused listening, musical shapes emerge from the sound mass. With the intention of musically assimilating into the sonic characteristics of the geyser, the score for seven pianists and electronics translates its harmonic, dynamic and rhythmic activity, from the lowest to highest register of the piano. Each pianist focuses on an octave and is projected through a speaker, which results in a diffused sound around the audience. The electronics present the geyser in its original form, divided into seven tracks, one track per octave, in order to be fused with the pianos and spatialized around the hall.

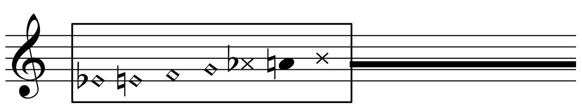
Performance notes

Time

The piece must be performed with a stopwatch. Each bar is 4 seconds long and always includes a time cue:



Pitch (categories of predominance)



Until a new box or silence suggest a change, perform the pitches from the box considering the following categories:

 High presence

 Medium presence

 Low presence

Rhythm (categories of saturation)

Each boxed pitch material should always be performed using either of the following rhythmic categories.
The symbol , indicates all rhythms performed must be <u>irregular</u>, never repeating pulses or rhythms):

 not more than one note per two seconds

 not more than one note per second

 not more than two notes per second

 not more than four notes per second

 not more than eight notes per second

 as many notes as possible per second

Technical requirements

There are several options for performance of this piece, considering the availability of resources in the performance space. The options range from an abundant resource scenario to a simple setup. The following is the setup used for the premier of the piece in Old Cabell Hall, at the University of Virginia, in October 2016.

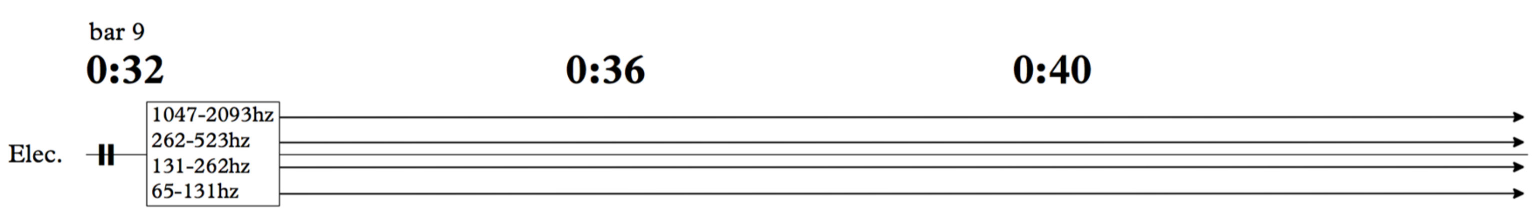
- 1 piano on stage; 3 pianos outside the concert hall, in practice rooms (sound proof) and telematically amplified in the hall.
- 7 microphones (assortment of cardioid condenser microphones, among them AKG 414 or similar)
- XLR cables
- Stage snake (minimum 8-channel input)
- 7 microphone stands
- 7 loudspeakers (placed surrounding the audience)
- Mixing board
- Laptop with a Digital Audio Workstation (Logic, Reaper, etc.)

Electronics

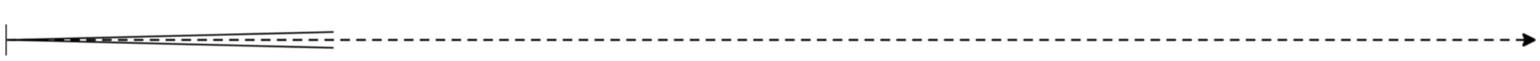
The electronics of the piece are seven tracks with seven frequency strata of the field recording of the geyser from which the piano parts are derived. Each track corresponds to the frequency range of each of the piano parts. The tracks are the following:

2093-4186hz
1047-2093hz
523-1047hz
262-523hz
131-262hz
65-131hz
20-65hz

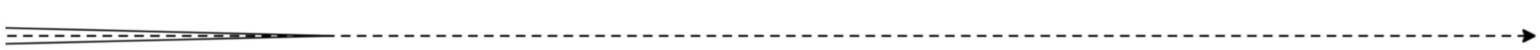
The electronic performer follows the score at the mixer board, fading the seven tracks in and out and balancing their dynamics with those of the pianists. <u>The dynamics of the geyser sounds and the live performers must always be balanced</u>. The main notations are the following:



Sounding frequency band tracks throughout until otherwise notated



Crescendo throughout the dotted arrow



Diminuendo throughout the dotted arrow until track fade out

*The electronic tracks will be provided for performance upon request.

Setup

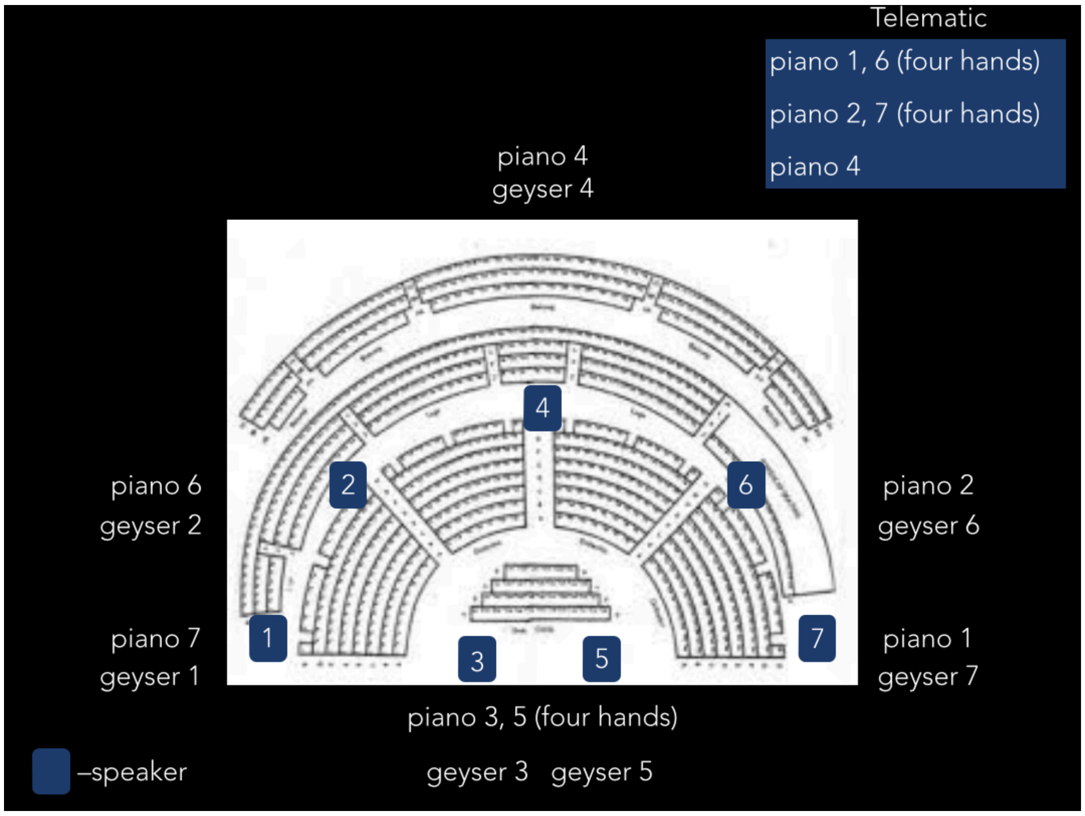
- Three of the four pianos are played four hands. The distribution of the parts are the following:

Piano parts 1 and 6	Offstage piano 1
Piano parts 2 and 7	Offstage piano 2
Piano part 4	Offstage piano 3
Piano parts 3 and 5	Stage piano

- Microphones must be as close to the strings as possible, in order to pick the minimum amount of sound from the pianist at the opposite end

- Routing of Piano parts and Electronics to loudspeakers (see diagram below):

Geyser 1 (track 1 of the electronics) / Piano 7	Speaker 1
Geyser 2 / Piano 6	Speaker 2
Geyser 3 / Piano 5	Speaker 3
Geyser 4 / Piano 4	Speaker 4
Geyser 5 / Piano 3	Speaker 5
Geyser 6 / Piano 2	Speaker 6
Geyser 7 / Piano 1	Speaker 7

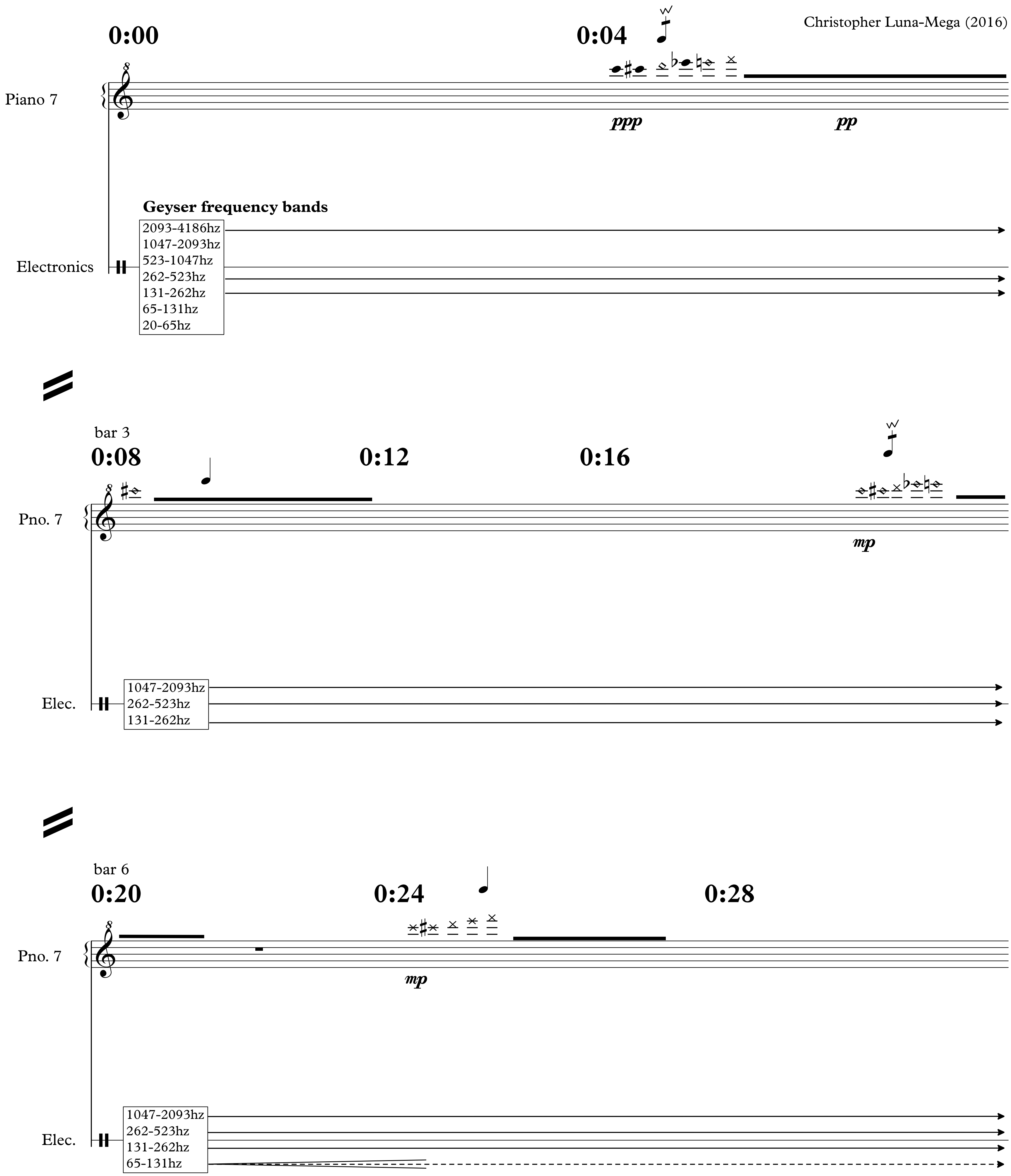


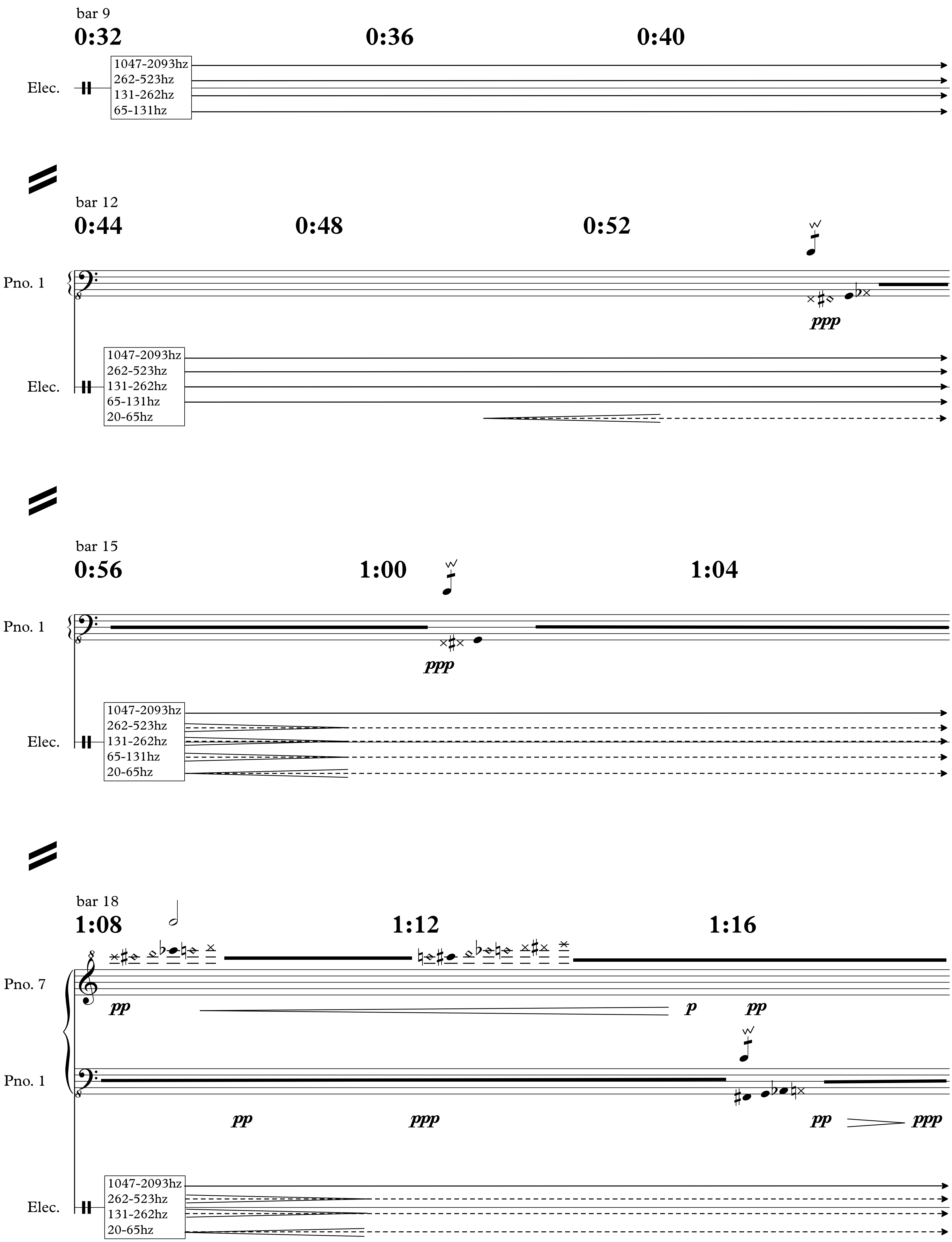
Full Score

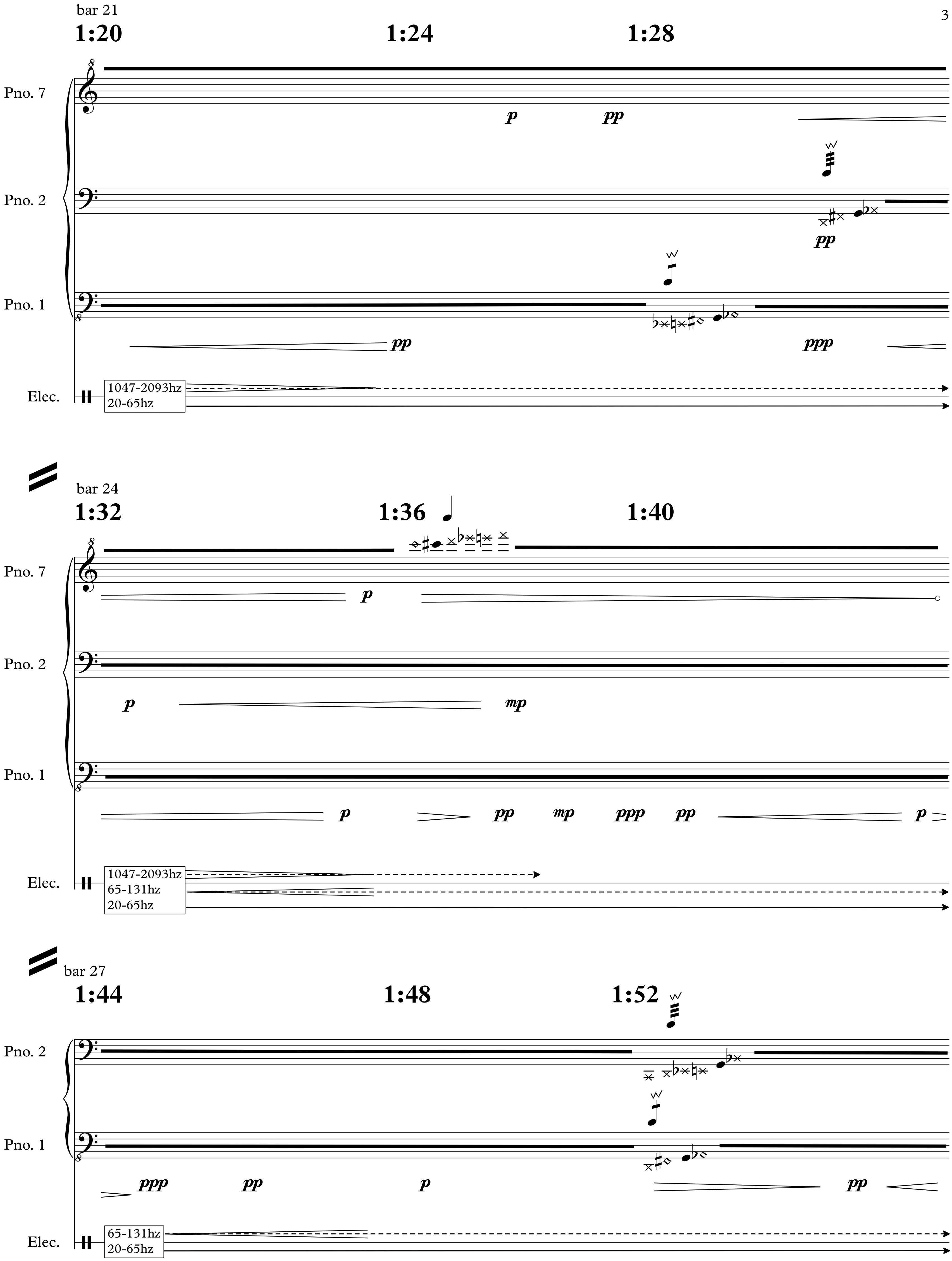
Geysir

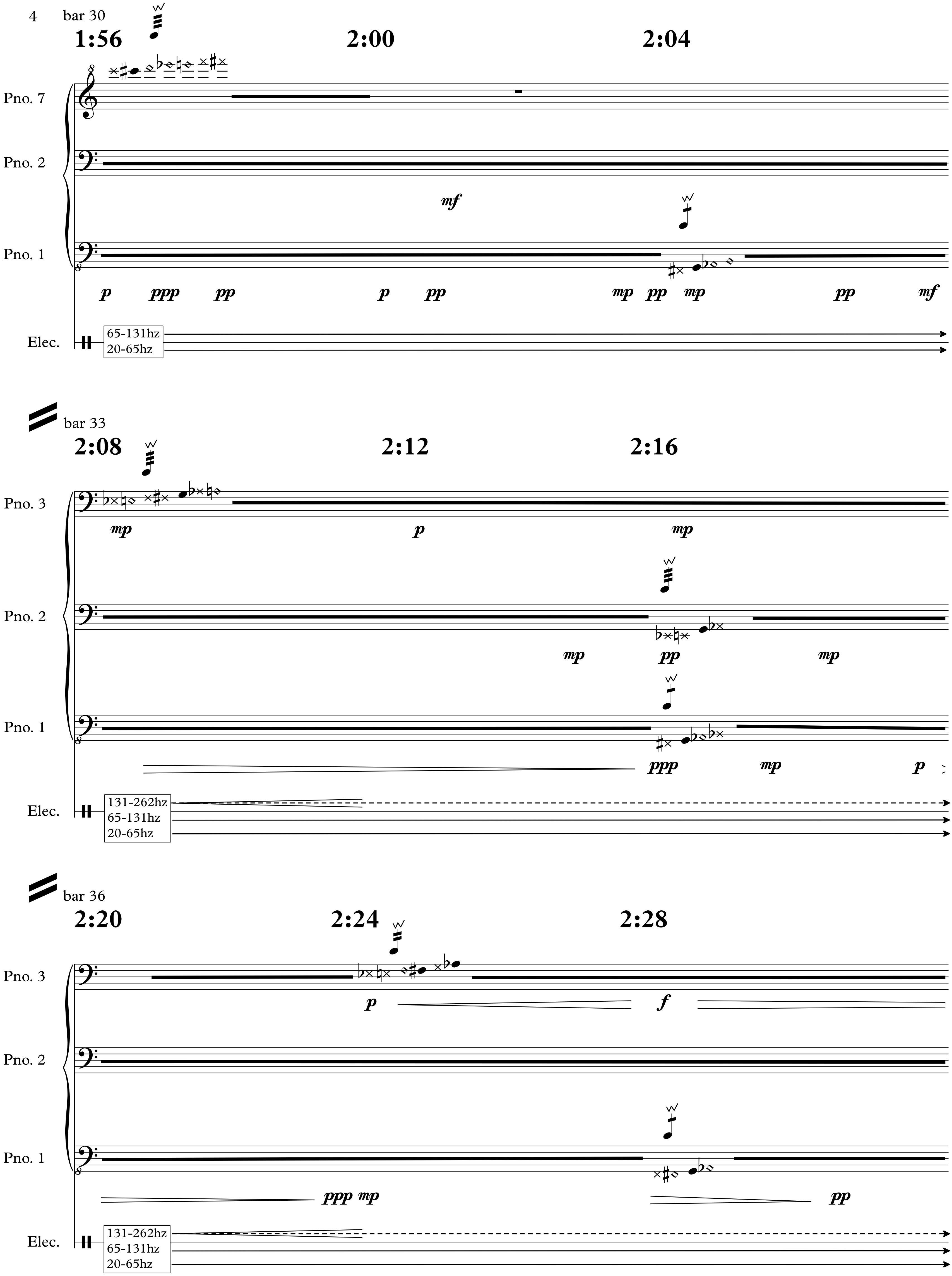
for seven pianists and electronics

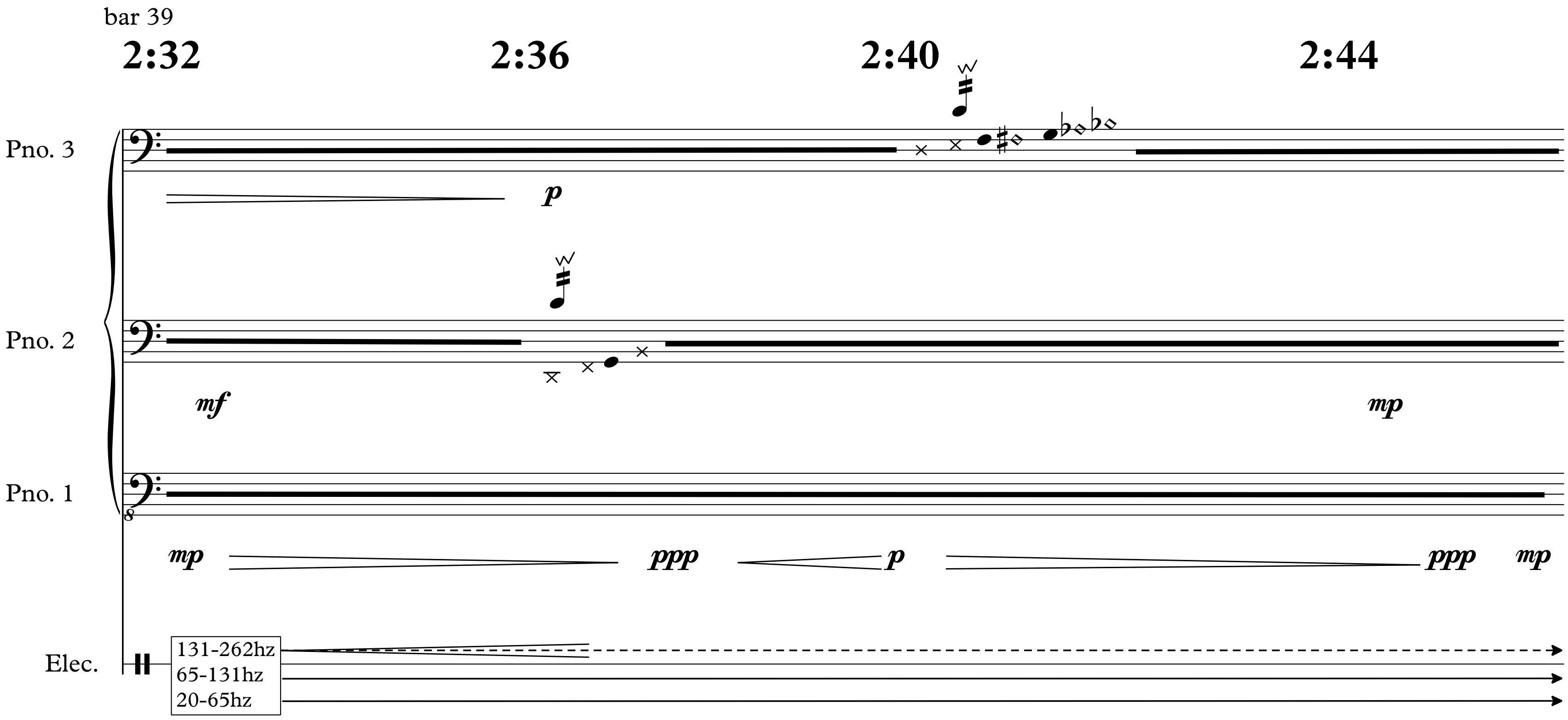
Christopher Luna-Mega (2016)

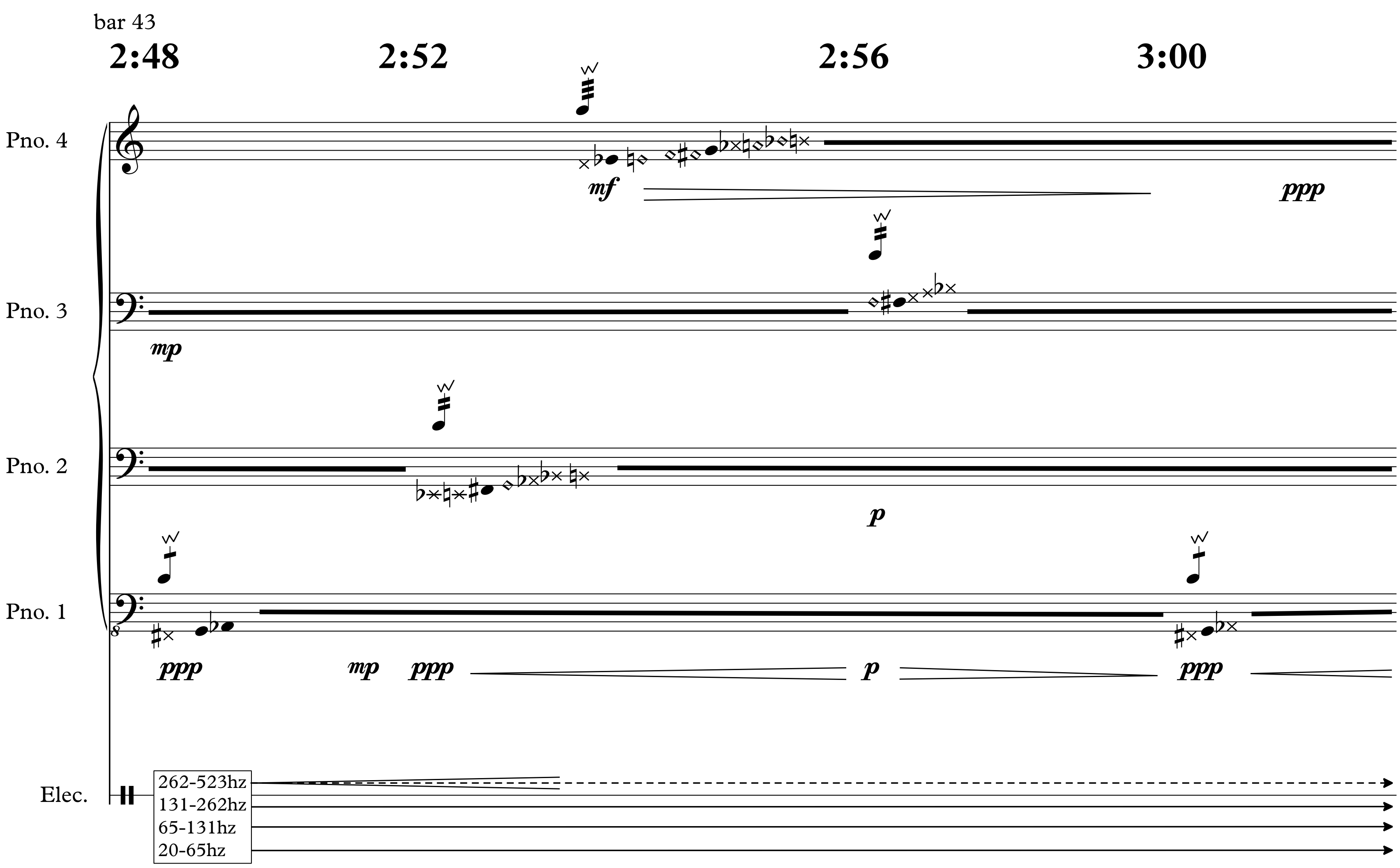


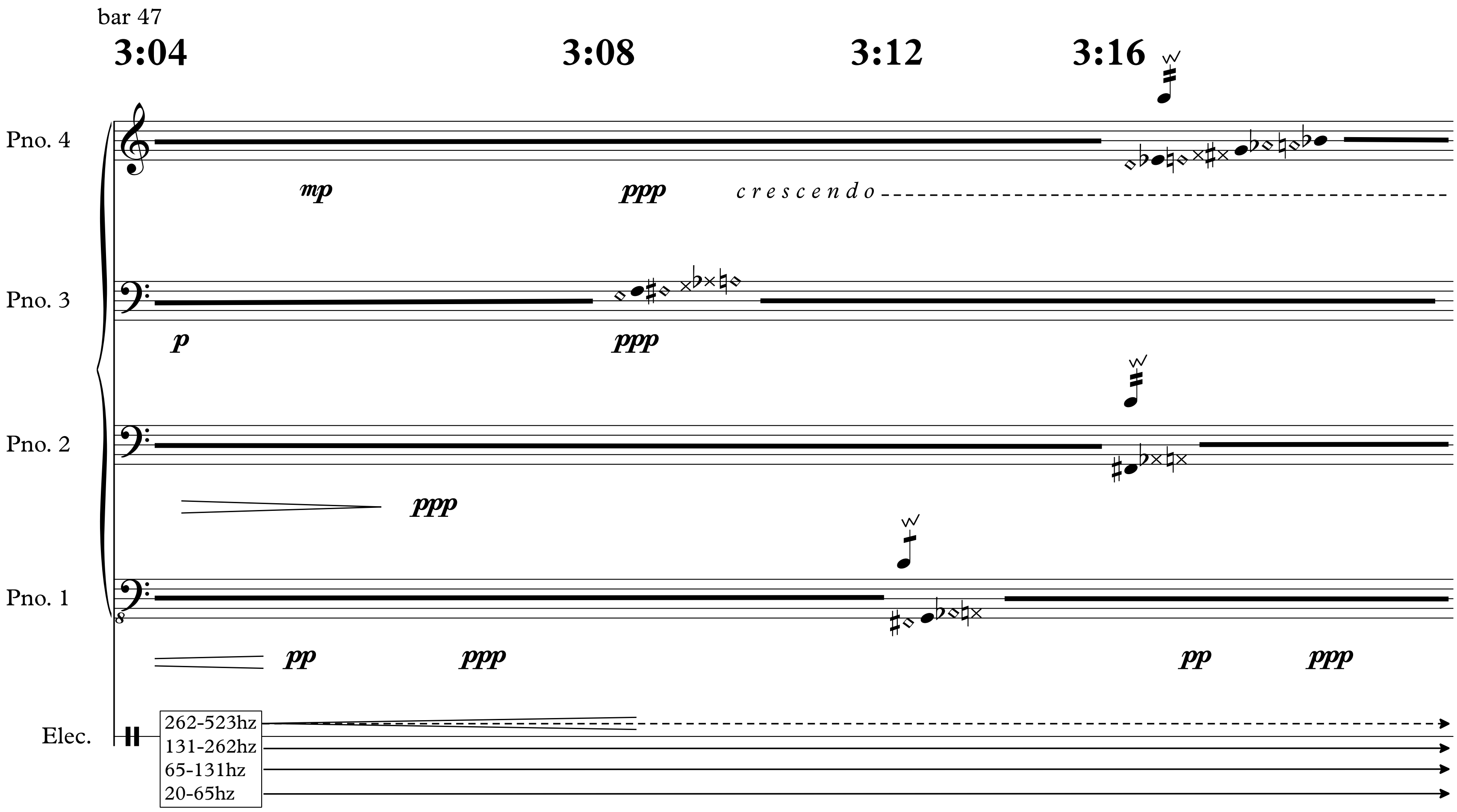

bar 9
0:32
0:36
0:40
Elec.
1047-2093hz
262-523hz
131-262hz
65-131hz
bar 12
0:44
0:48
0:52
Pno. 1
ppp
Elec.
1047-2093hz
262-523hz
131-262hz
65-131hz
20-65hz
bar 15
0:56
1:00
1:04
Pno. 1
ppp
Elec.
1047-2093hz
262-523hz
131-262hz
65-131hz
20-65hz
bar 18
1:08
1:12
1:16
Pno. 7
pp
p
pp
Pno. 1
pp
ppp
pp
ppp
Elec.
1047-2093hz
262-523hz
131-262hz
20-65hz

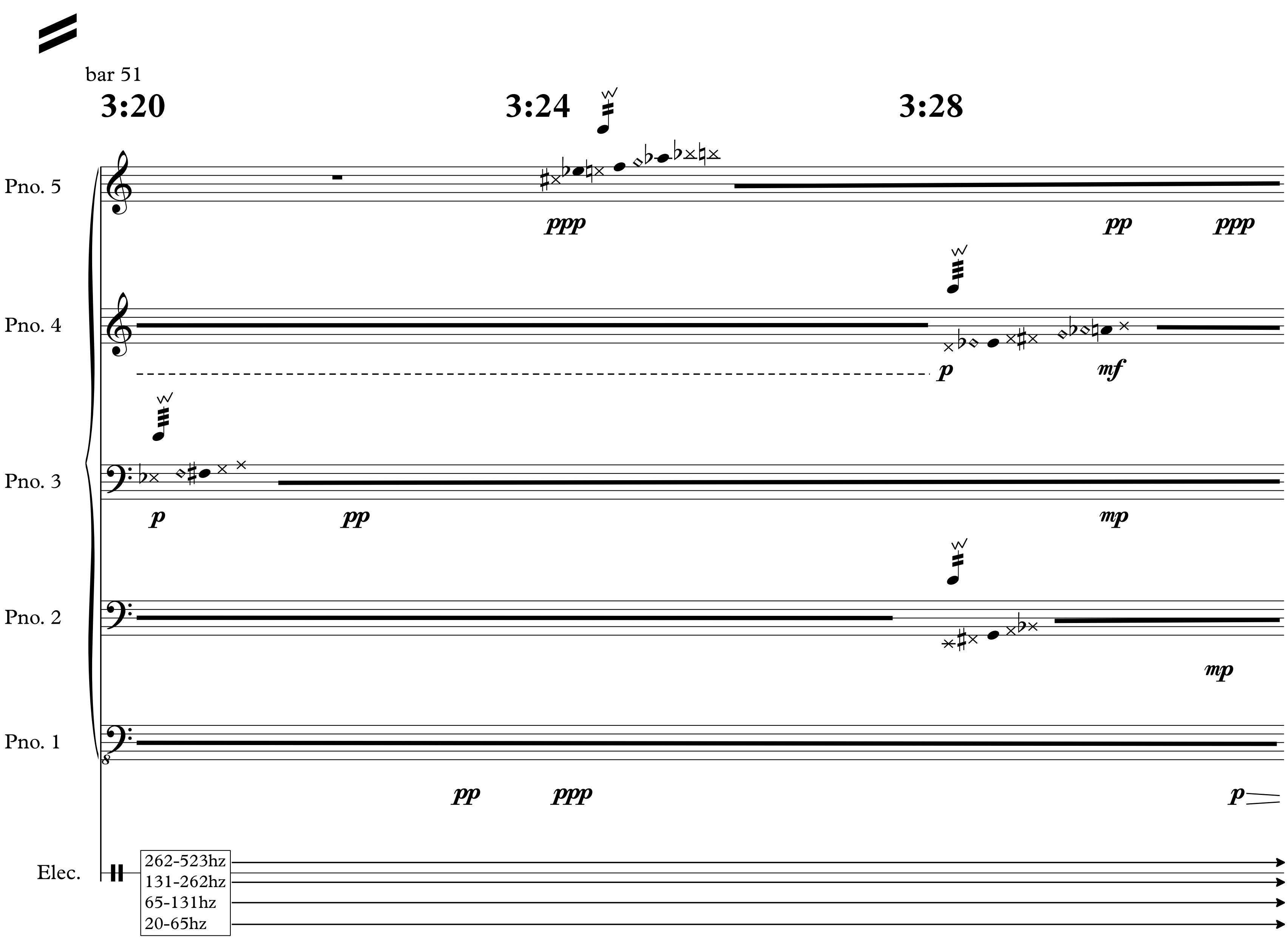

bar 21
1:20
1:24
1:28
Pno. 7
p
pp
Pno. 2
pp
Pno. 1
pp
ppp
Elec.
1047-2093hz
20-65hz
bar 24
1:32
1:36
1:40
Pno. 7
p
Pno. 2
p
mp
Pno. 1
p
pp
mp
ppp
pp
p
Elec.
1047-2093hz
65-131hz
20-65hz
bar 27
1:44
1:48
1:52
Pno. 2
Pno. 1
ppp
pp
p
pp
Elec.
65-131hz
20-65hz

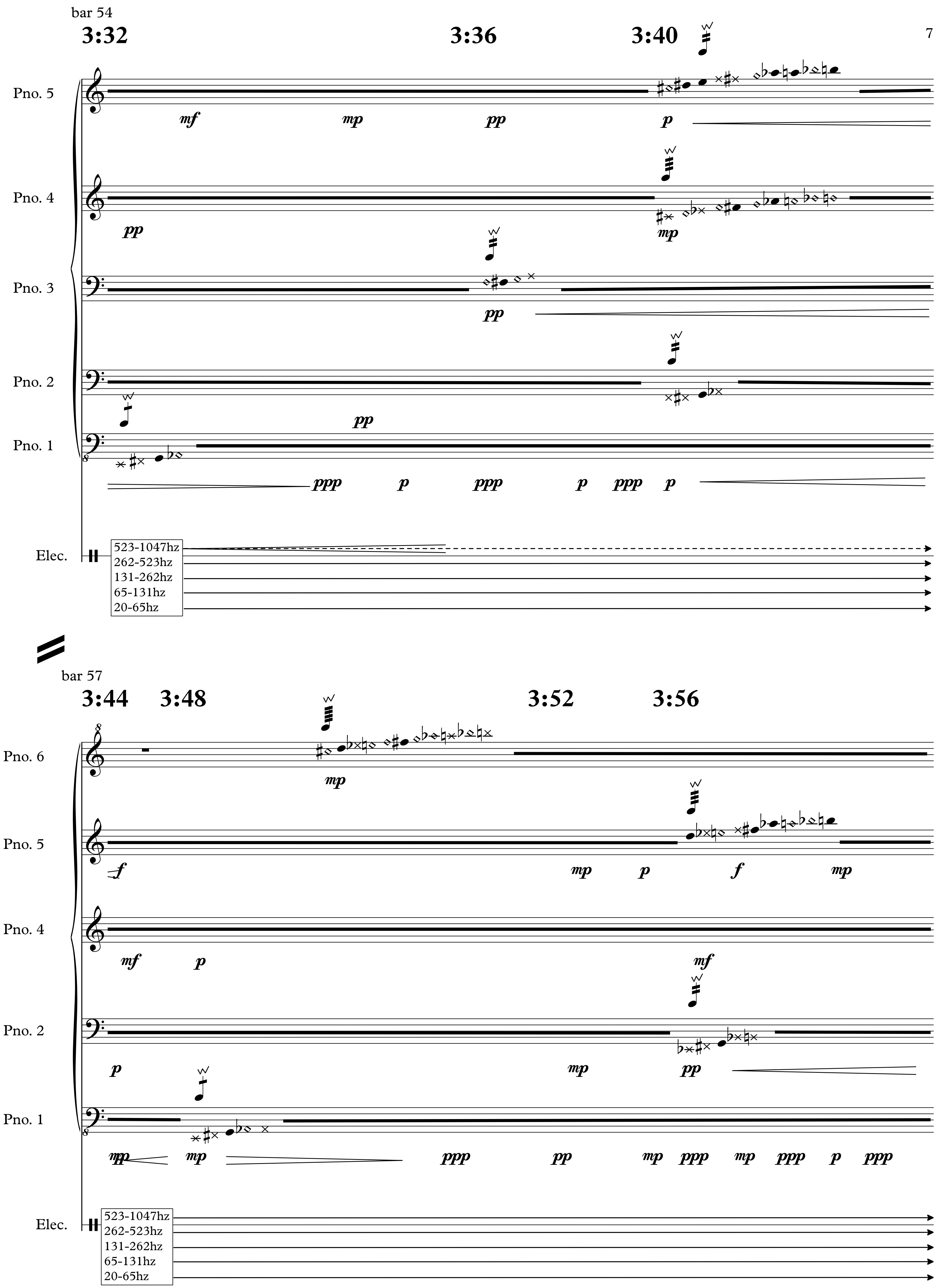

bar 30
1:56
2:00
2:04
Pno. 7
Pno. 2
Pno. 1
Elec.
65-131hz
20-65hz
bar 33
2:08
2:12
2:16
Pno. 3
Pno. 2
Pno. 1
Elec.
131-262hz
65-131hz
20-65hz
bar 36
2:20
2:24
2:28
Pno. 3
Pno. 2
Pno. 1
Elec.
131-262hz
65-131hz
20-65hz

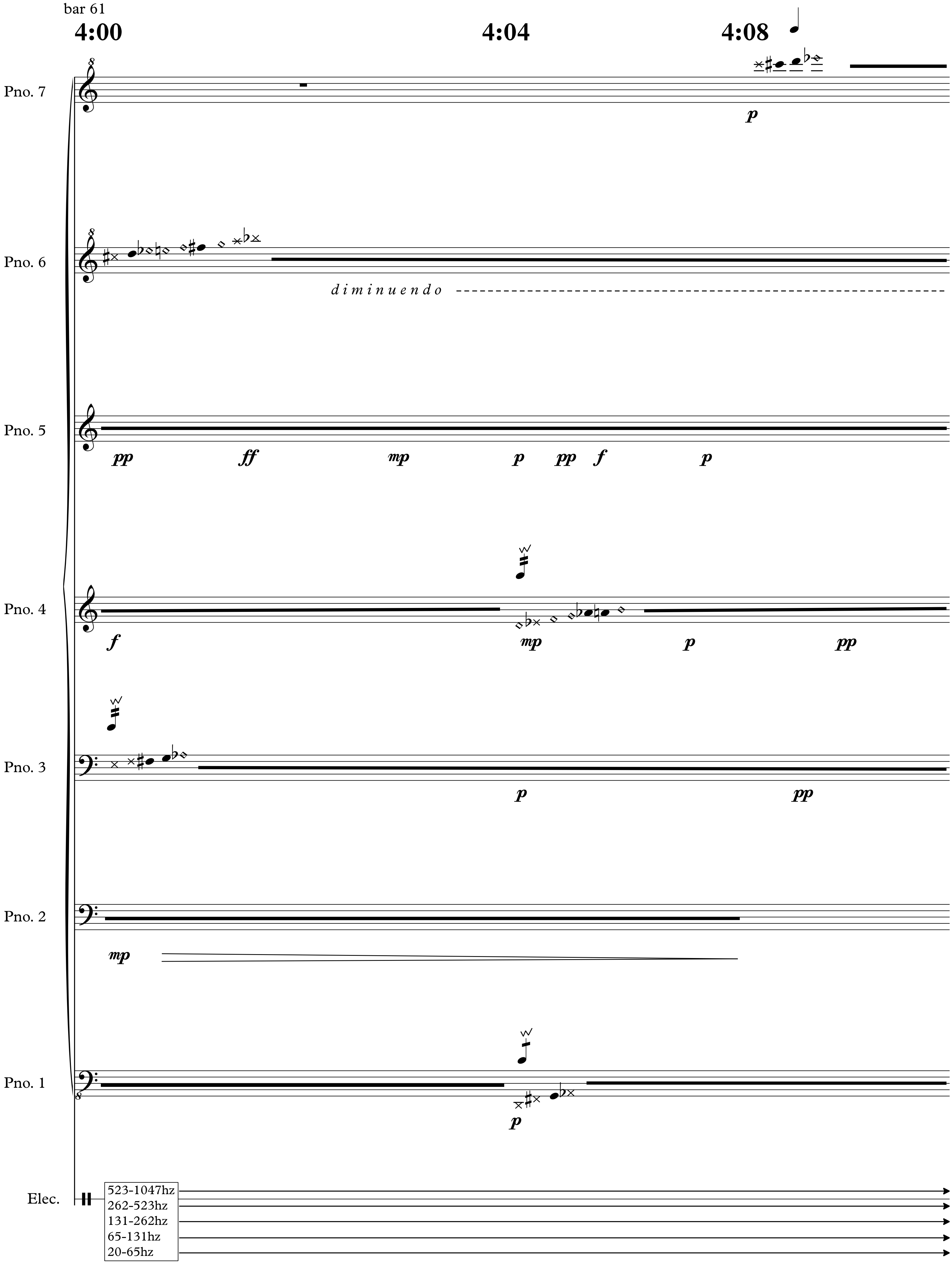

bar 39
2:32
2:36
2:40
2:44
Pno. 3
p
Pno. 2
mf
mp
Pno. 1
mp
ppp
p
ppp
mp
Elec.
131-262hz
65-131hz
20-65hz


bar 43
2:48
2:52
2:56
3:00
Pno. 4
mf
ppp
Pno. 3
mp
Pno. 2
p
Pno. 1
ppp
mp
ppp
p
ppp
Elec.
262-523hz
131-262hz
65-131hz
20-65hz


bar 47
3:04
3:08
3:12
3:16
Pno. 4
Pno. 3
Pno. 2
Pno. 1
Elec.
mp
ppp
crescendo
p
ppp
ppp
pp
ppp
pp
ppp
262-523hz
131-262hz
65-131hz
20-65hz


bar 51
3:20
3:24
3:28
Pno. 5
Pno. 4
Pno. 3
Pno. 2
Pno. 1
Elec.
ppp
pp
ppp
p
mf
p
pp
mp
mp
pp
ppp
p
262-523hz
131-262hz
65-131hz
20-65hz


bar 54
3:32
3:36
3:40
Pno. 5
mf mp pp p
Pno. 4
pp
mp
Pno. 3
pp
Pno. 2
pp
Pno. 1
ppp p ppp p ppp p
Elec.
523-1047hz
262-523hz
131-262hz
65-131hz
20-65hz
bar 57
3:44
3:48
3:52
3:56
Pno. 6
mp
Pno. 5
f
mp p f mp
Pno. 4
mf p
mf
Pno. 2
p
mp
pp
Pno. 1
mp mp ppp pp mp ppp mp ppp p ppp
Elec.
523-1047hz
262-523hz
131-262hz
65-131hz
20-65hz


bar 61
4:00
4:04
4:08
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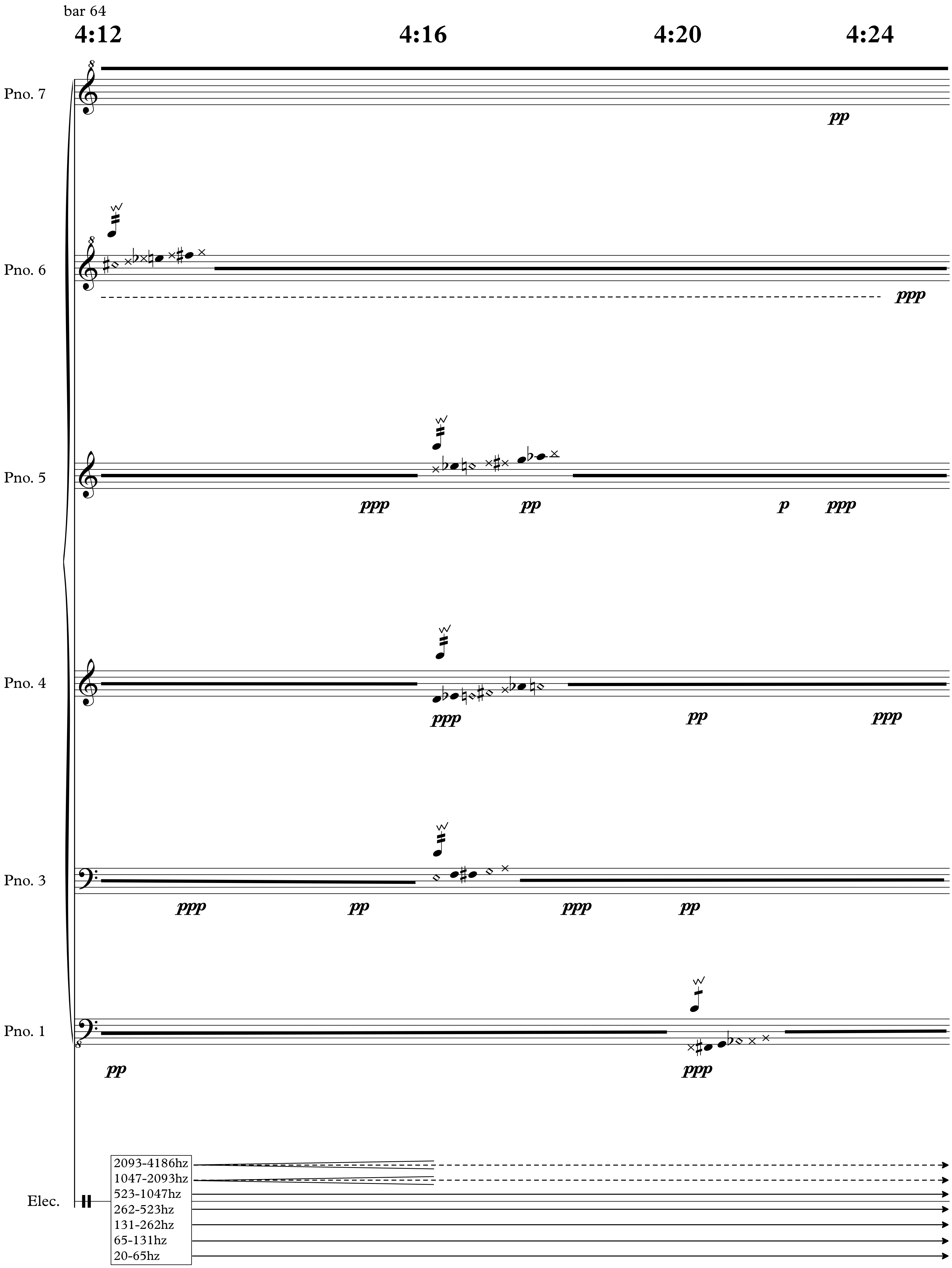
p
Pno. 6
diminuendo
Pno. 5
pp
ff
mp
p
pp
f
p
Pno. 4
f
mp
p
pp
Pno. 3
p
pp
Pno. 2
mp
Pno. 1
p
Elec.
523-1047hz
262-523hz
131-262hz
65-131hz
20-65hz


bar 64
4:12
4:16
4:20
4:24
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pp
Pno. 6
ppp
Pno. 5
ppp
pp
p
ppp
Pno. 4
ppp
pp
ppp
Pno. 3
ppp
pp
ppp
pp
Pno. 1
pp
ppp
Elec.
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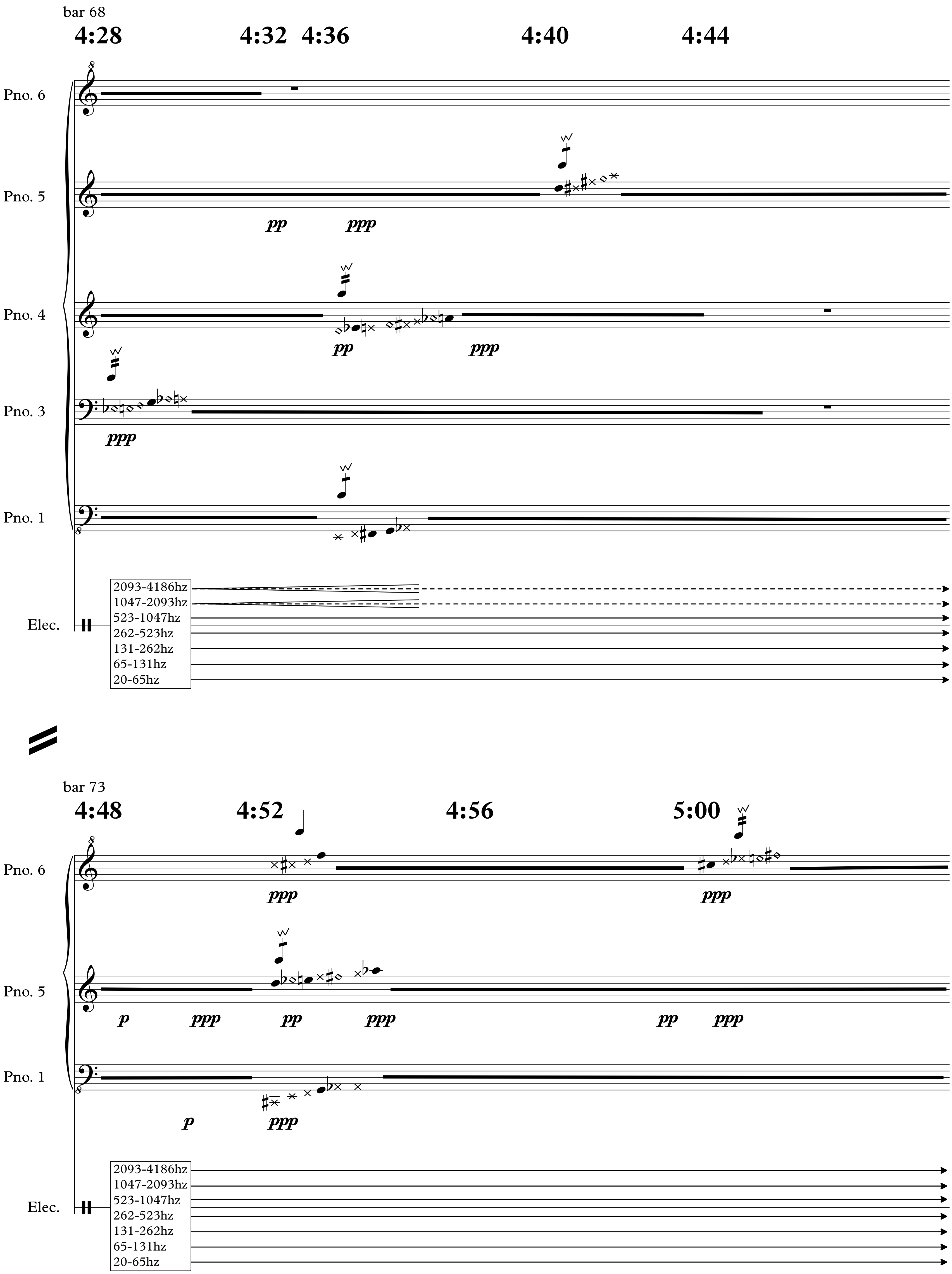
523-1047hz
262-523hz
131-262hz
65-131hz
20-65hz


bar 68
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4:40
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Pno. 6
Pno. 5
pp
ppp
Pno. 4
pp
ppp
Pno. 3
ppp
Pno. 1
Elec.
2093-4186hz
1047-2093hz
523-1047hz
262-523hz
131-262hz
65-131hz
20-65hz
bar 73
4:48
4:52
4:56
5:00
Pno. 6
ppp
ppp
Pno. 5
p
ppp
pp
ppp
pp
ppp
Pno. 1
p
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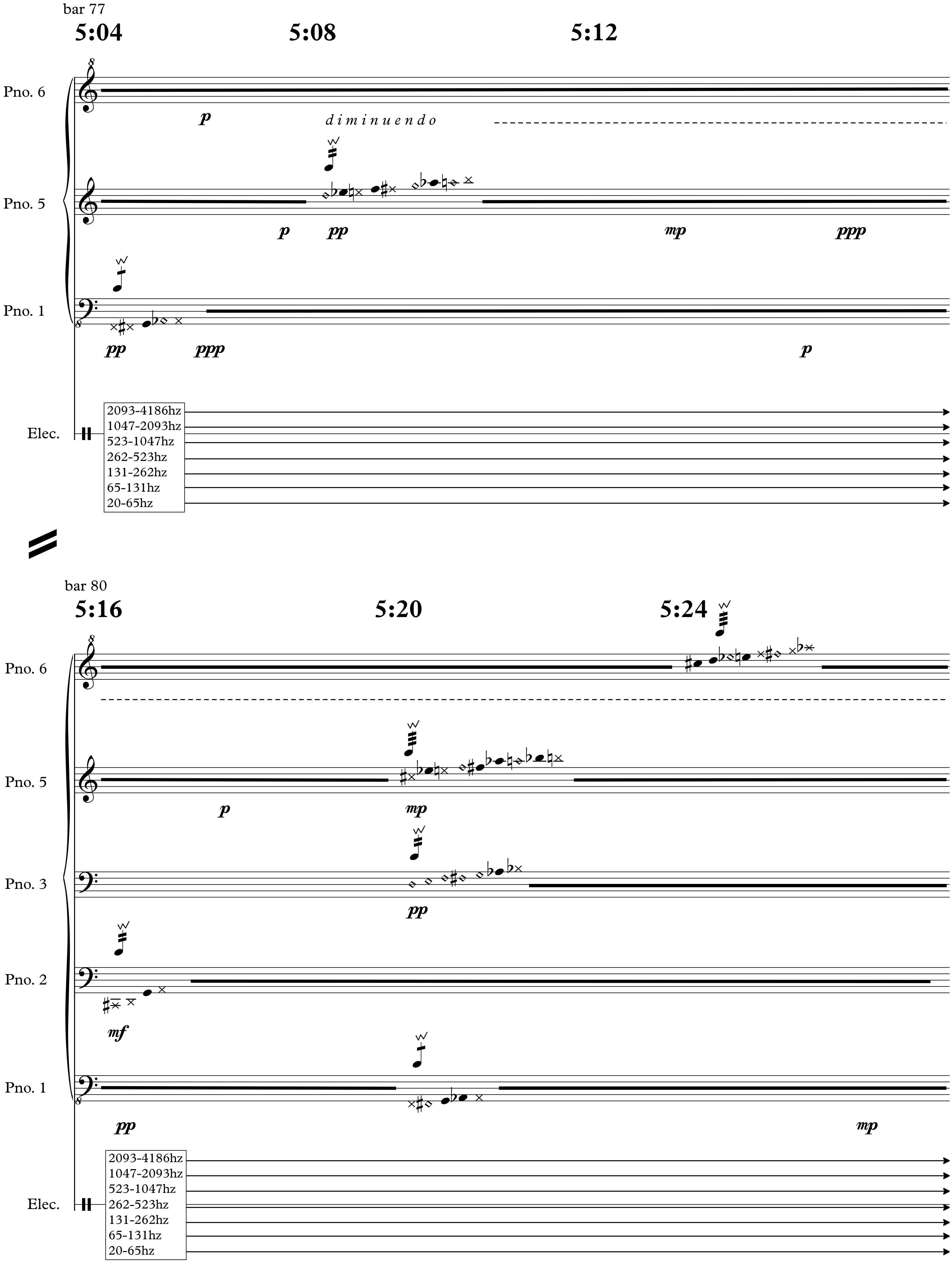
bar 77
5:04
5:08
5:12
Pno. 6
p
diminuendo
Pno. 5
p
pp
mp
ppp
Pno. 1
pp
ppp
p
Elec.
2093-4186hz
1047-2093hz
523-1047hz
262-523hz
131-262hz
65-131hz
20-65hz
bar 80
5:16
5:20
5:24
Pno. 6
Pno. 5
p
mp
Pno. 3
pp
Pno. 2
mf
Pno. 1
pp
mp
Elec.
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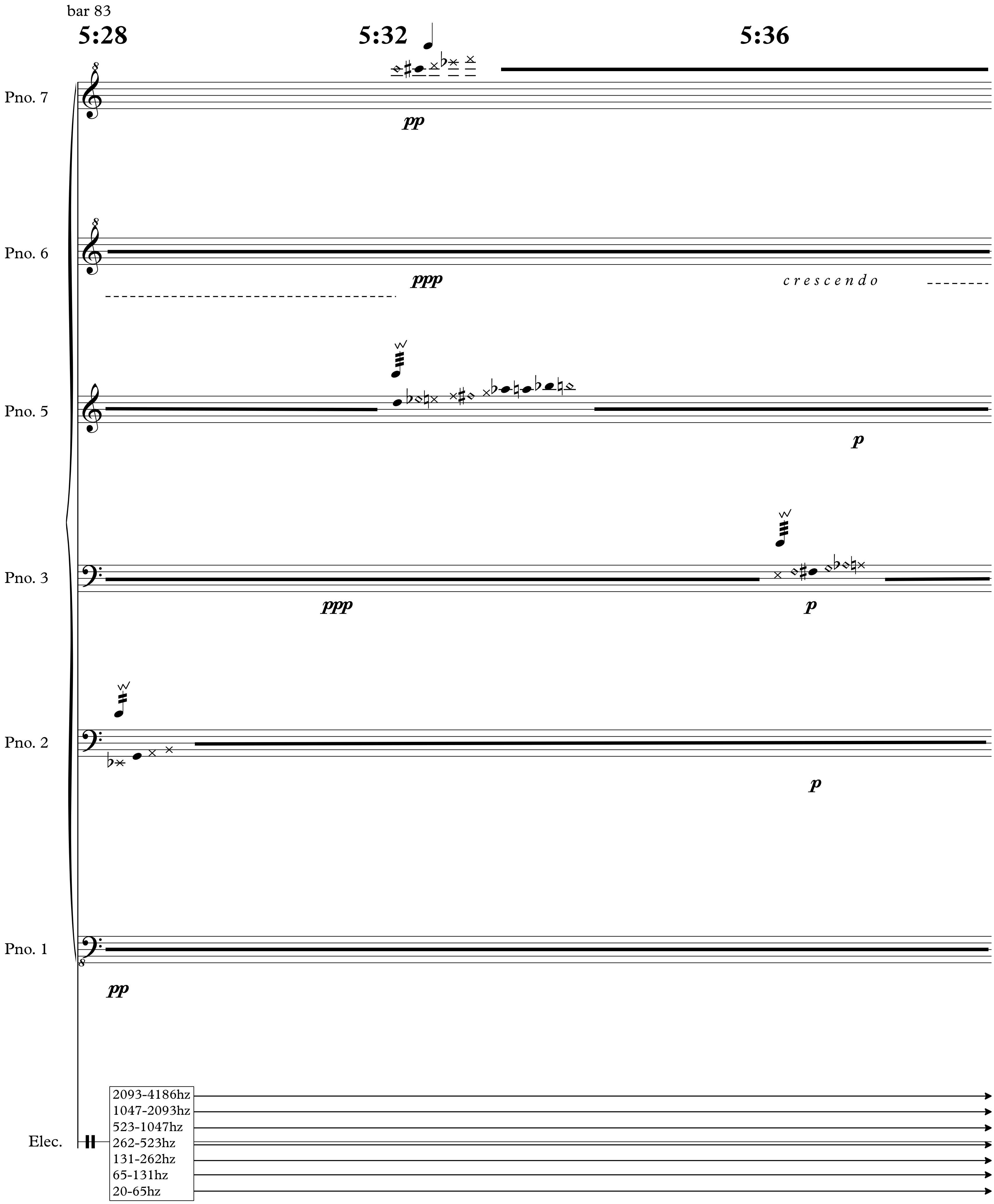
65-131hz
20-65hz


bar 83
5:28
5:32
5:36
Pno. 7
pp
Pno. 6
ppp
crescendo
Pno. 5
p
Pno. 3
ppp
p
Pno. 2
p
Pno. 1
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2093-4186hz
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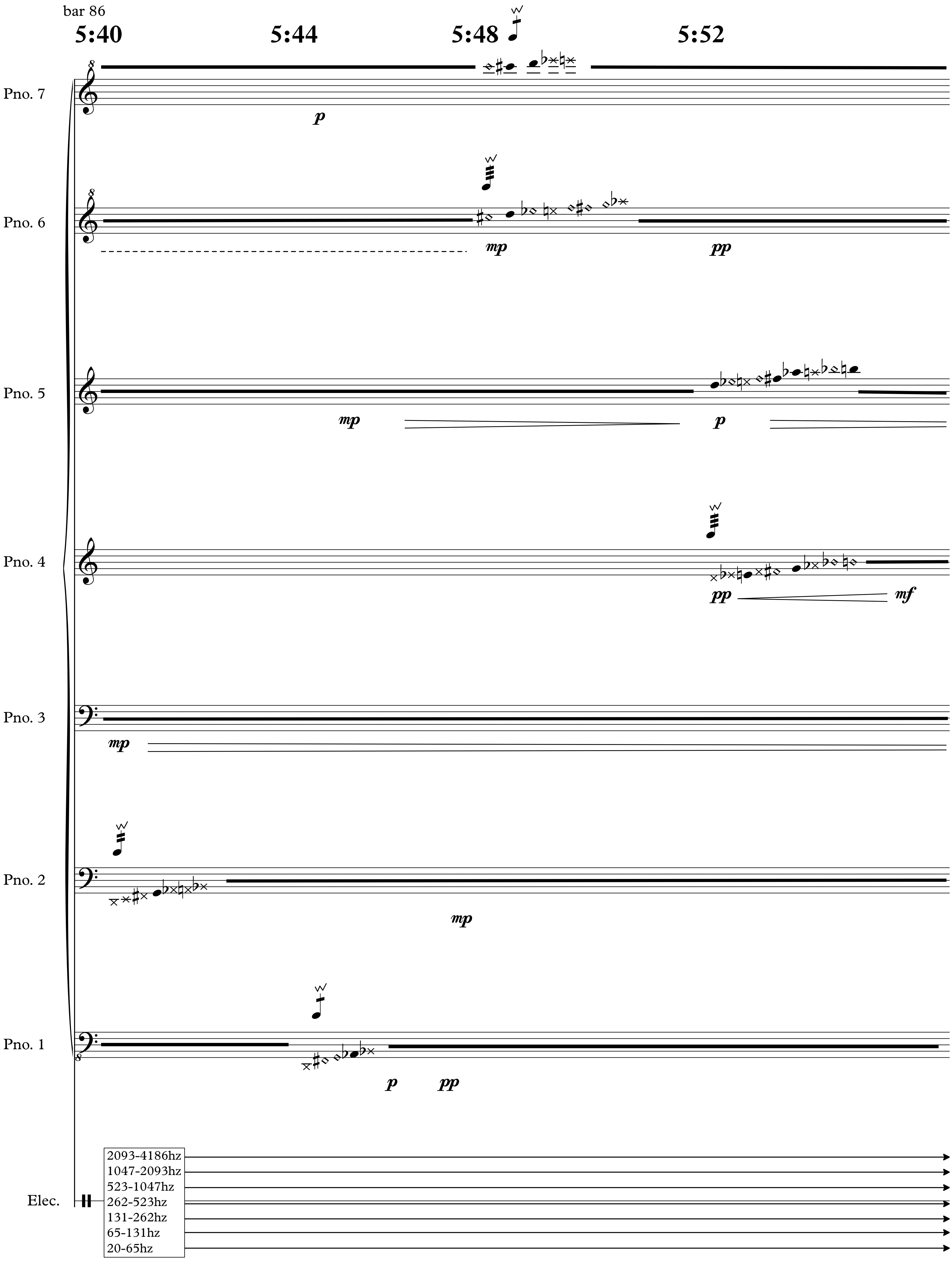
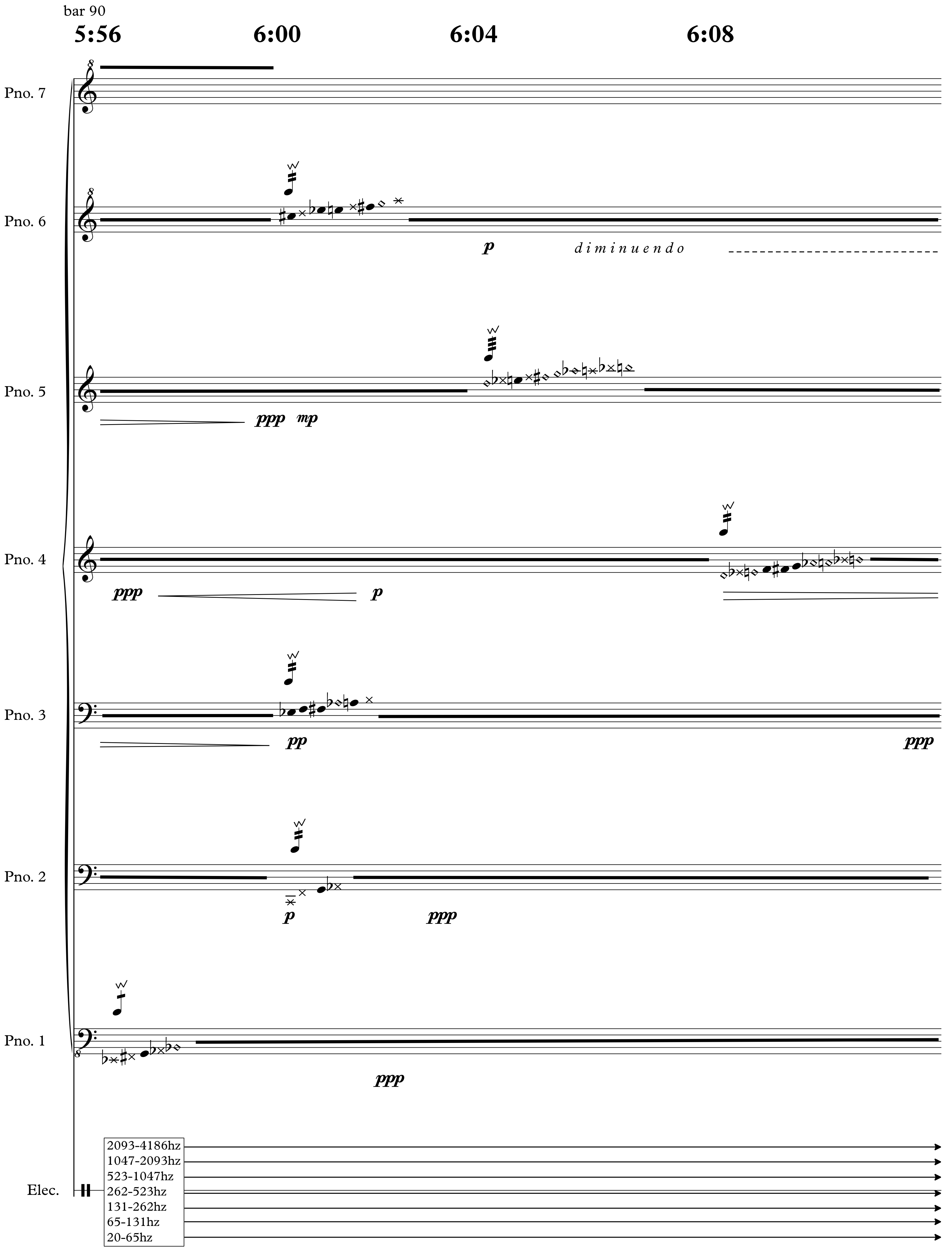
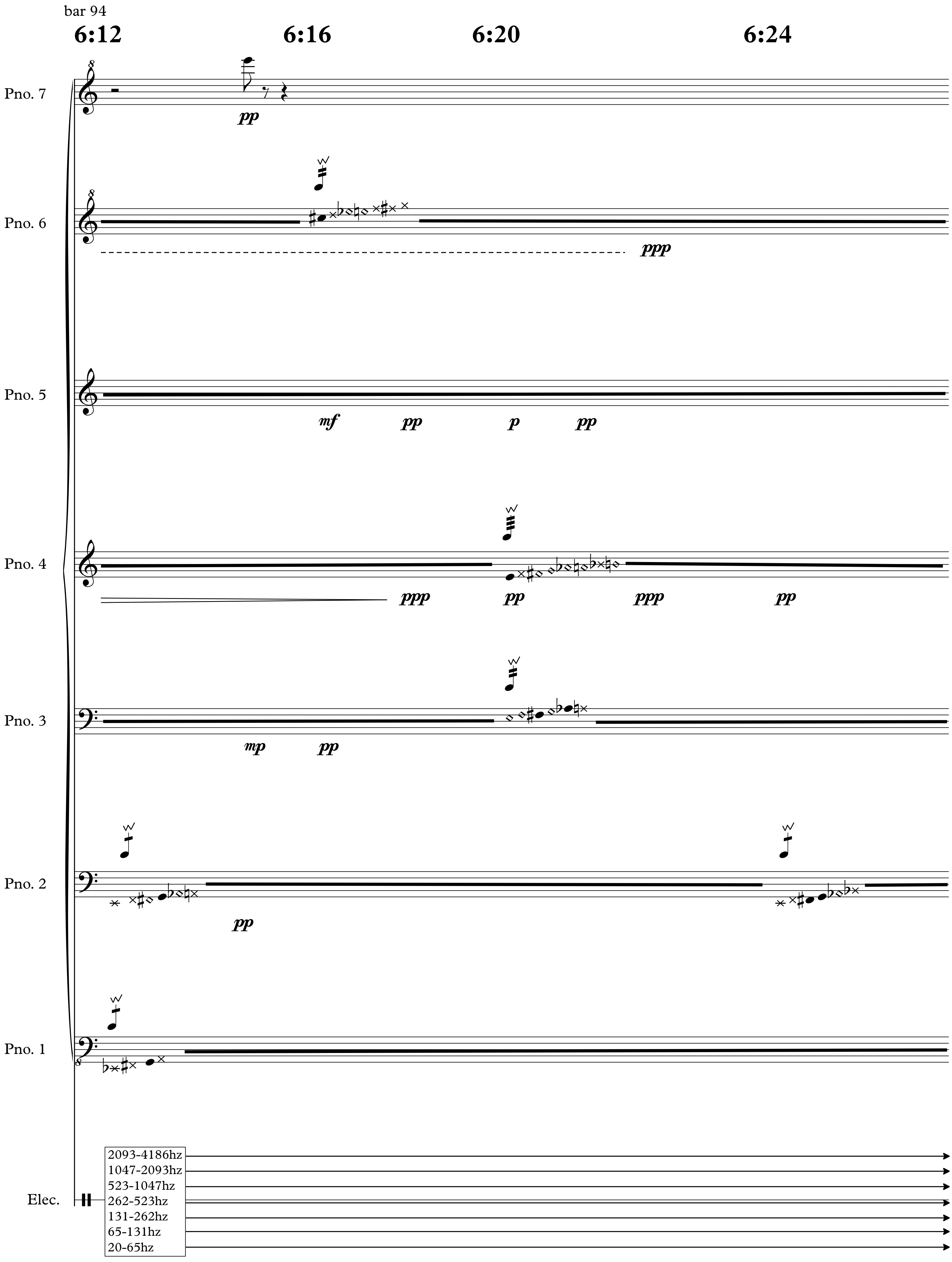
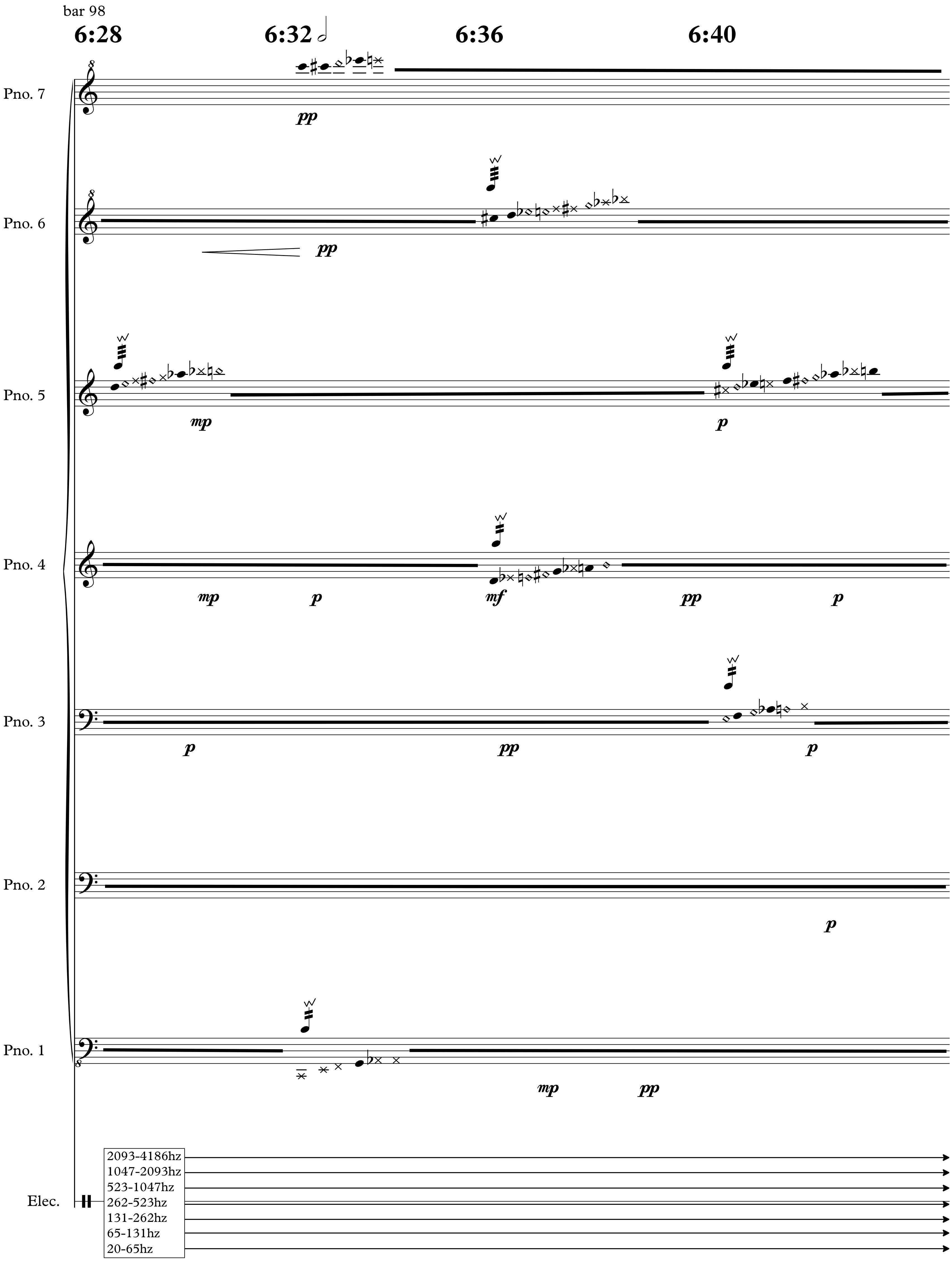
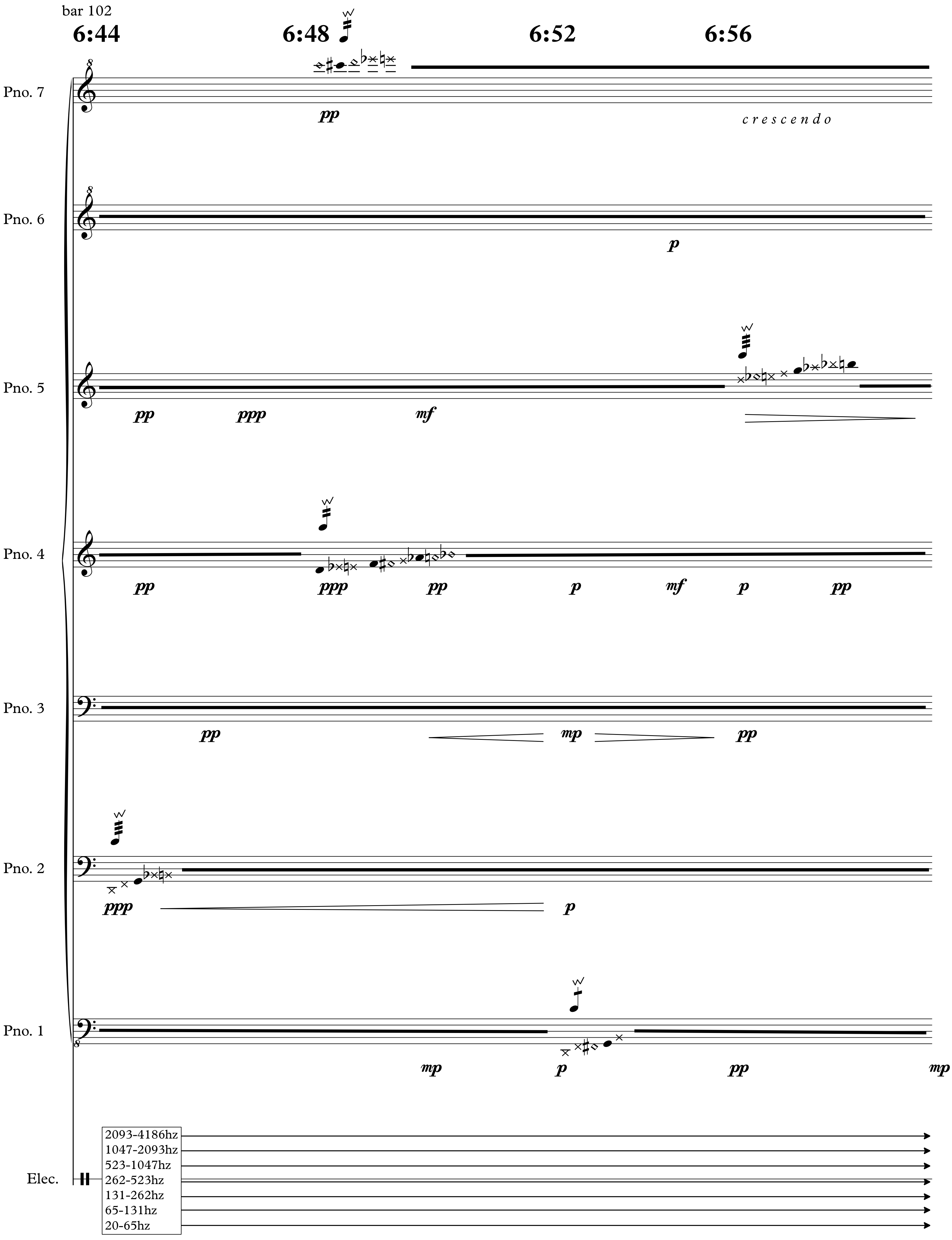
65-131hz
20-65hz

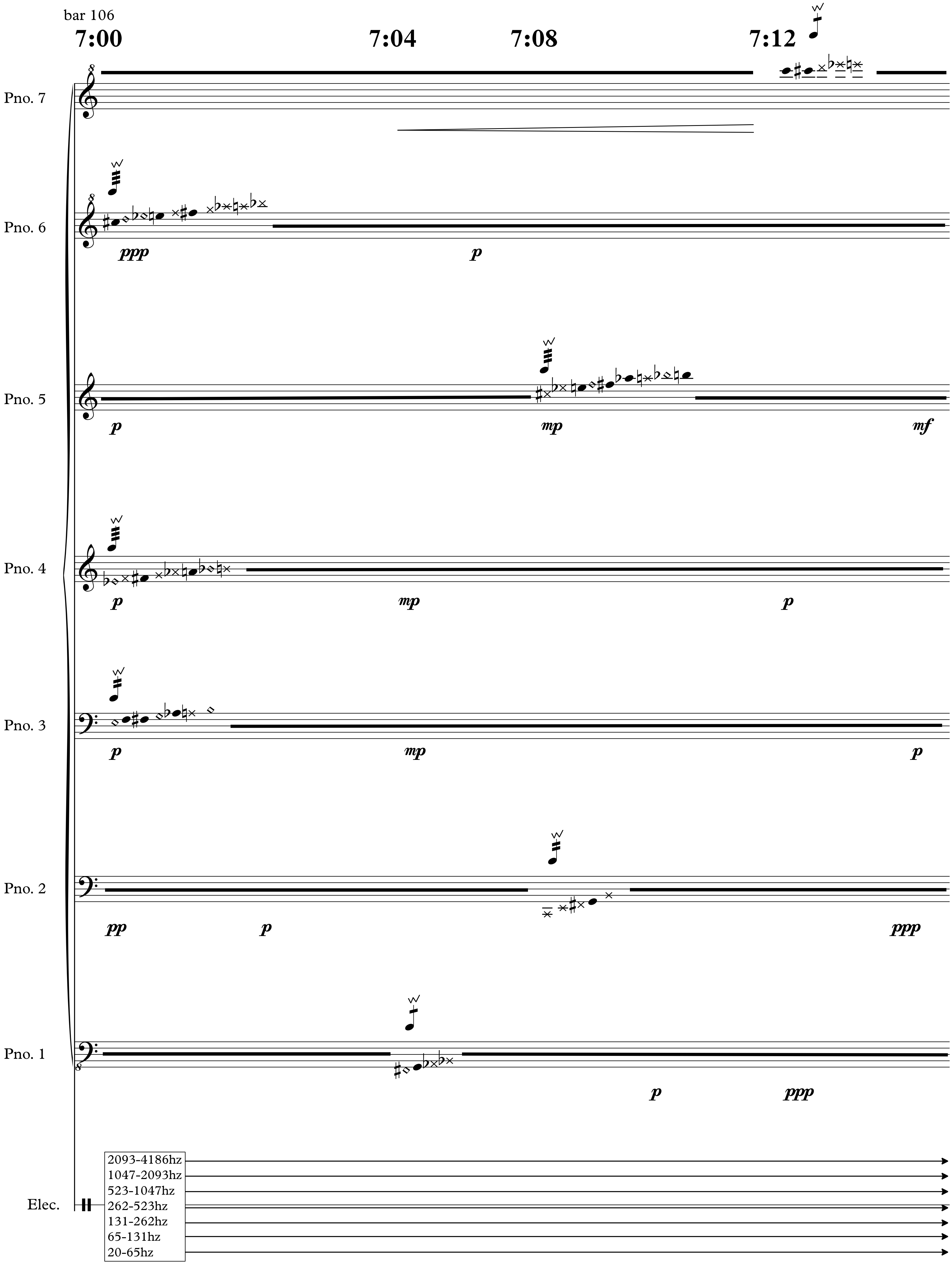

bar 86
5:40
5:44
5:48
5:52
Pno. 7
p
Pno. 6
mp
pp
Pno. 5
mp
p
Pno. 4
pp
mf
Pno. 3
mp
Pno. 2
mp
Pno. 1
p
pp
Elec.
2093-4186hz
1047-2093hz
523-1047hz
262-523hz
131-262hz
65-131hz
20-65hz


bar 90
5:56
6:00
6:04
6:08
Pno. 7
Pno. 6
p
diminuendo
Pno. 5
ppp
mp
Pno. 4
ppp
p
Pno. 3
pp
ppp
Pno. 2
p
ppp
Pno. 1
ppp
Elec.
2093-4186hz
1047-2093hz
523-1047hz
262-523hz
131-262hz
65-131hz
20-65hz


bar 94
6:12
6:16
6:20
6:24
Pno. 7
pp
Pno. 6
ppp
Pno. 5
mf
pp
p
pp
Pno. 4
ppp
pp
ppp
pp
Pno. 3
mp
pp
Pno. 2
pp
Pno. 1
Elec.
2093-4186hz
1047-2093hz
523-1047hz
262-523hz
131-262hz
65-131hz
20-65hz


bar 98
6:28
6:32
6:36
6:40
Pno. 7
pp
Pno. 6
pp
Pno. 5
mp
p
Pno. 4
mp
p
mf
pp
p
Pno. 3
p
pp
p
Pno. 2
p
Pno. 1
mp
pp
Elec.
2093-4186hz
1047-2093hz
523-1047hz
262-523hz
131-262hz
65-131hz
20-65hz


bar 102
6:44
6:48
6:52
6:56
Pno. 7
pp
crescendo
Pno. 6
p
Pno. 5
pp
ppp
mf
Pno. 4
pp
ppp
pp
p
mf
p
pp
Pno. 3
pp
mp
pp
Pno. 2
ppp
p
Pno. 1
mp
p
pp
mp
Elec.
2093-4186hz
1047-2093hz
523-1047hz
262-523hz
131-262hz
65-131hz
20-65hz


bar 106
7:00
7:04
7:08
7:12
Pno. 7
Pno. 6
ppp
p
Pno. 5
p
mp
mf
Pno. 4
p
mp
p
Pno. 3
p
mp
p
Pno. 2
pp
p
ppp
Pno. 1
p
ppp
Elec.
2093-4186hz
1047-2093hz
523-1047hz
262-523hz
131-262hz
65-131hz
20-65hz


bar 110
7:16
8:40
Pno. 7
subito
Pno. 6
subito
Pno. 5
subito
Pno. 4
subito
Pno. 3
subito
Pno. 2
subito
Pno. 1
subito
Elec.
2093-4186hz
1047-2093hz
523-1047hz
262-523hz
131-262hz
65-131hz
20-65hz

Waves Break Aural Shores

Portrait of Puerto Marqués

for saxophone quartet and electronics

Christopher Luna-Mega

Performance notes

The score combines traditional notation with proportional notations. Written in 4/4 at ♩ = 60, each bar is four seconds long.

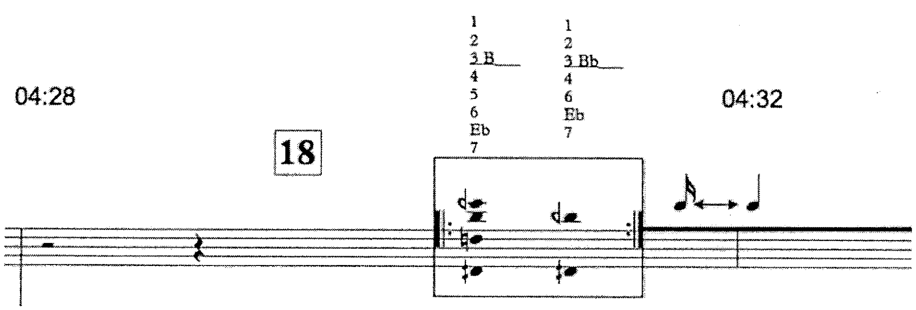
Types of notation

1. Traditional notation combined with proportional glissandi

The pitches written with rhythmic values serve as points of departure and arrival of the glissandi. Noteheads intersected by glissandi (i.e. A, B𝄳 and A𝄲, bar 2) should not be articulated and only indicate a reference pitch for the glissandi. Glissandi lines serve as legato slurs.

Glissandi may be interpreted proportionally and must be understood as a general indication of a gesture rather than a precise description of contour. However, it is important to privilege microtonal rather than chromatic pitch space.

2. Boxed sound(s) within repeat signs, extended with a horizontal beam



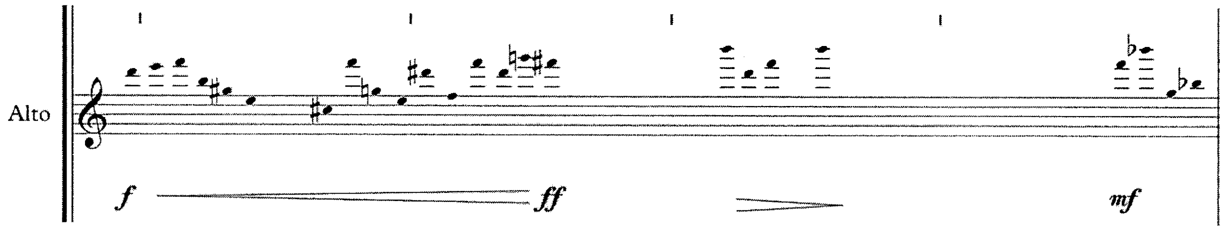
The boxed sound(s) may appear at any place within the four beats of the bar, either with a precise indication of the location in the bar or placed proportionally within the bar, less precisely.

The bold horizontal beam right of the box indicates the span of time that the material repeats and varies.

The durations for the sounds are notated right of the box. In the example, the durations for either of the two sounds should vary between a ♪ and a ♩ , until there is a new duration specified or a change in material.

Pauses between repetitions must be as short as possible. When breathing, make the smallest possible pause that is comfortable between repetitions.

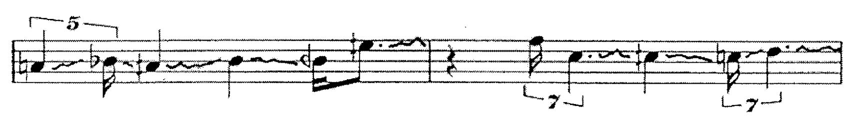
3. Proportionally placed smaller noteheads



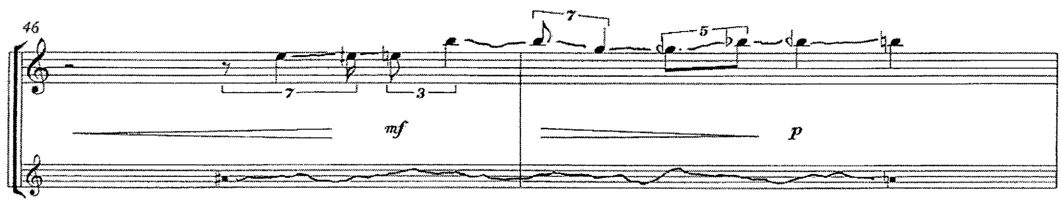
Stemless and smaller size noteheads are distributed throughout the four beats of the bar, which are represented by small markers above the staff. Articulations are optional to the performer.

Techniques

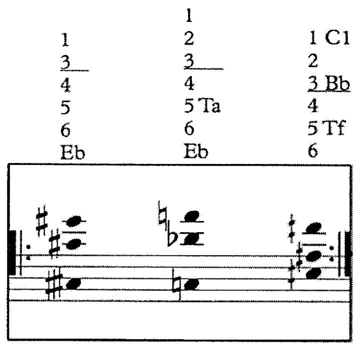
Aeolic (air) **sounds.** These may be produced with or without the mouthpiece depending on the time between sections using this technique and the rest of the sections. Choice of fingered pitch for the air sound is optional to the performer.



Glissandi. Move between notes by altering the embouchure (varying the pressure applied with the lips) or changing fingering depending on the interval for the glissando, instrument and register. In sections with glissandi, all written pitches followed by glissandi must be in the most continuous microtonal motion possible.



Polyphonic Glissandi. Traditional notes on the upper staff are played on the saxophone; square/stemless noteheads on the lower staff are sung and written proportionally. Pitches and pace of the glissandi are flexible for the performer depending on her/his singing skills.



Multiphonics. The notation and selection of multiphonics is based on *Le Sons Multiples Aux Saxophone*, by Daniel Kientzy, ed. Salabert. It may be provided upon request. When more than one multiphonic is provided, the performer must use all the options available throughout the given section. Refer to the "Boxed sound(s) within repeat signs" section for more information on the notation.

Electronics / Technical requirements

Waves Break Aural Shores (portrait of Puerto Marqués) is originally conceived for saxophone quartet and fixed media. There is an alternate version for saxophone quartet, fixed media and live processing. The technical requirements for the two versions using electronics are the following.

1. Saxophone quartet and fixed media

- Laptop and Digital Audio Workstation such as Logic, Reaper, etc.

- 2 loudspeakers (stereo)

- Field recording audio file included with score and parts. Track is synchronized with the time cues in the score and parts. Fixed media and performers' stopwatches must start at the same time.

2. Saxophone quartet, fixed media and electronics

- All fixed media requirements

- MacBook Pro laptop with OS 10.11, computer program Max 7.2

- Max patch for *Waves Break Aural Angles (portrait of Puerto Marqués)*

- Audio interface (MOTU 828 or similar) with 2-4 outs; optional mixing board

- Saxophone quartet amplification: 4 condenser microphones such as/similar to Shure SM-81

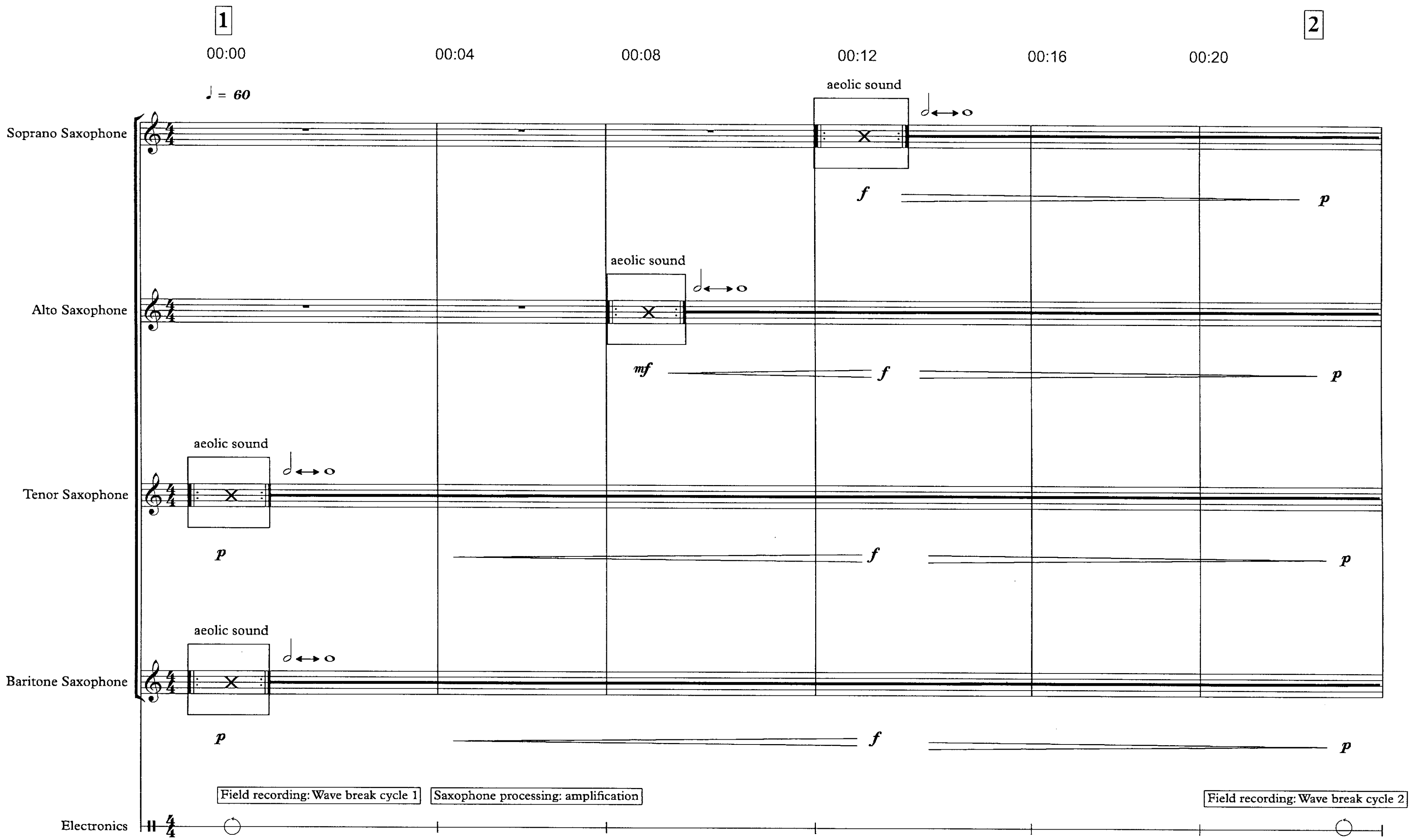
Waves Break Aural Shores

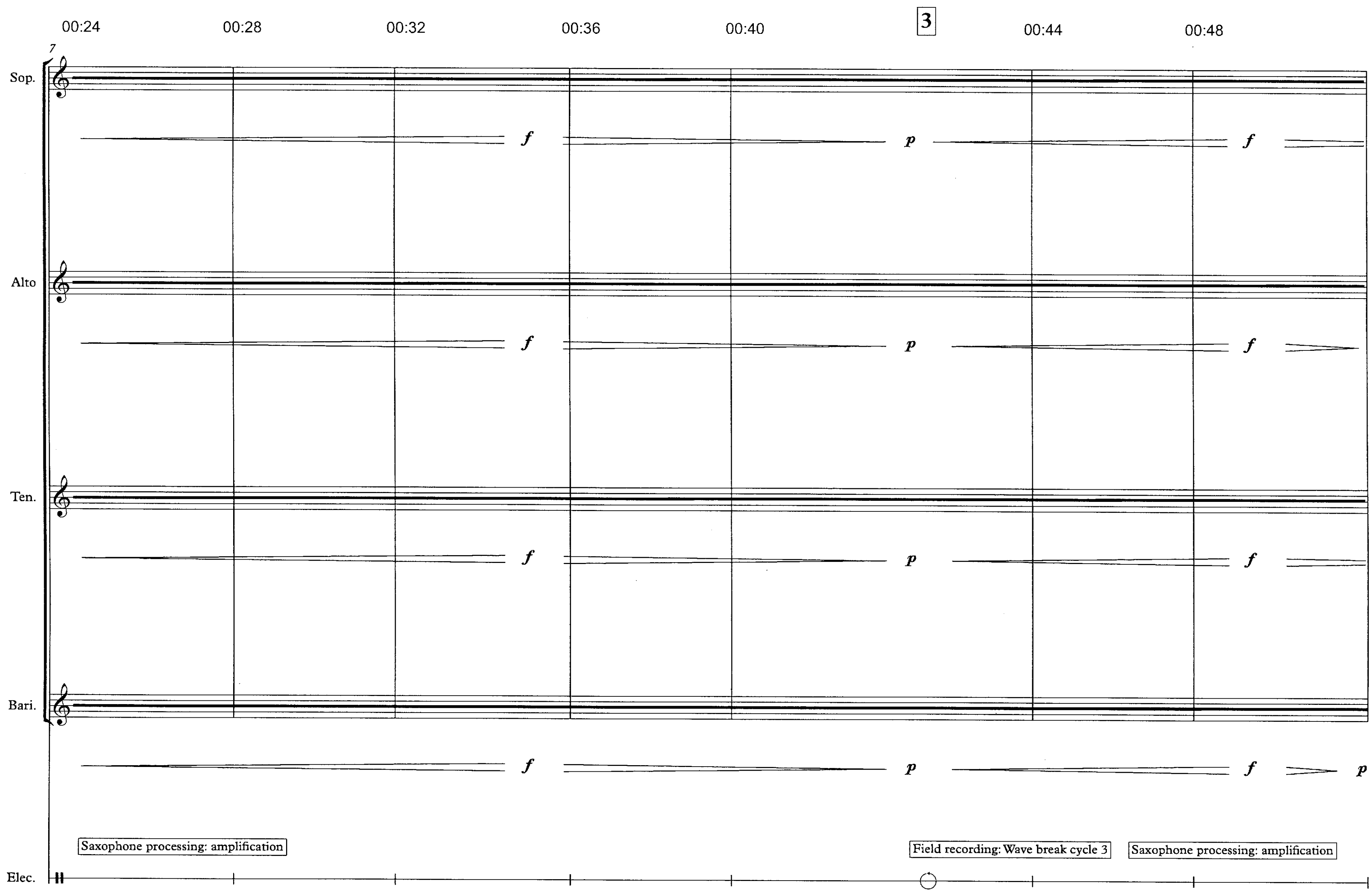
Portrait of Puerto Marqués

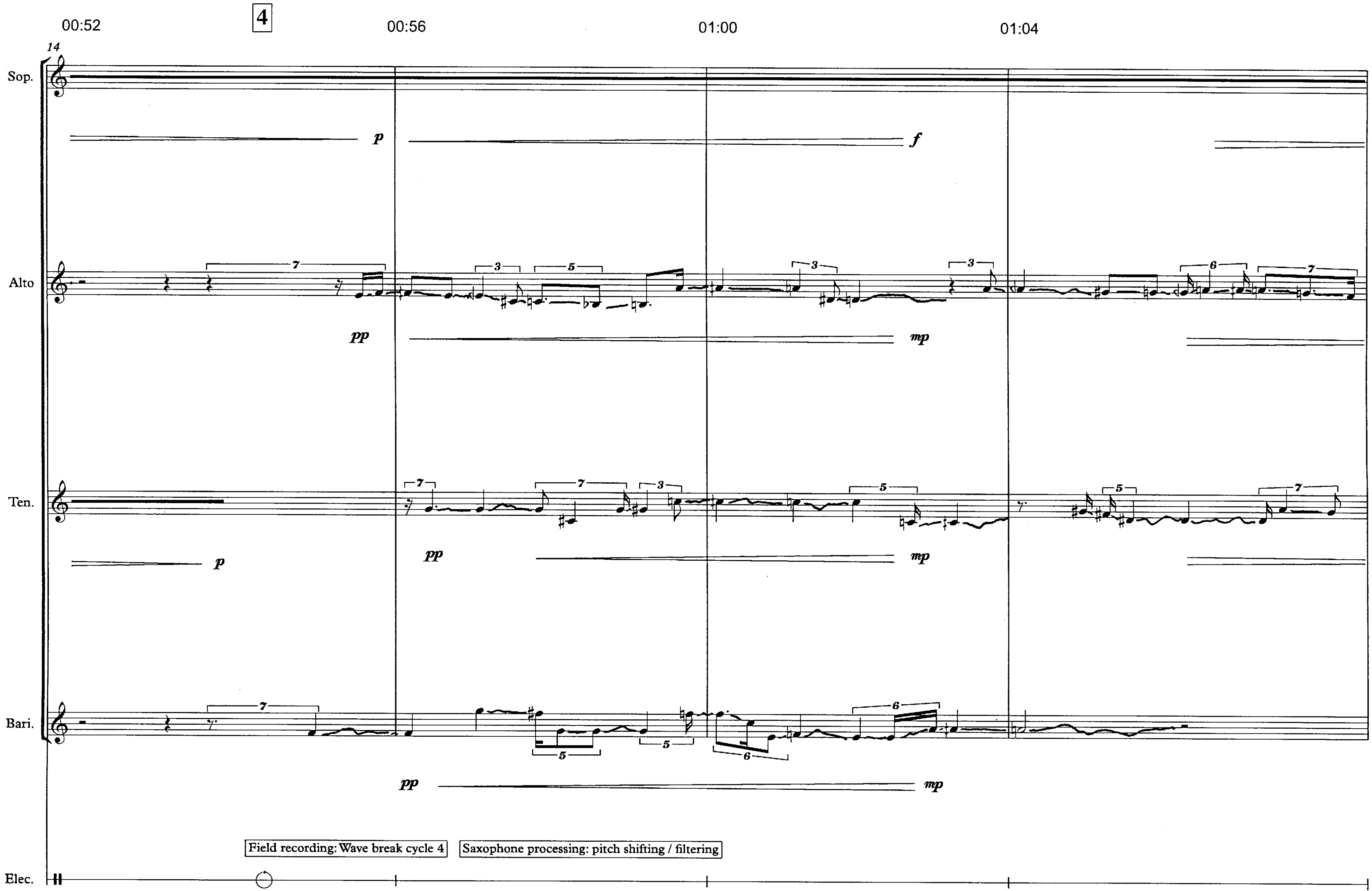
for saxophone quartet and electronics

Transposed Score

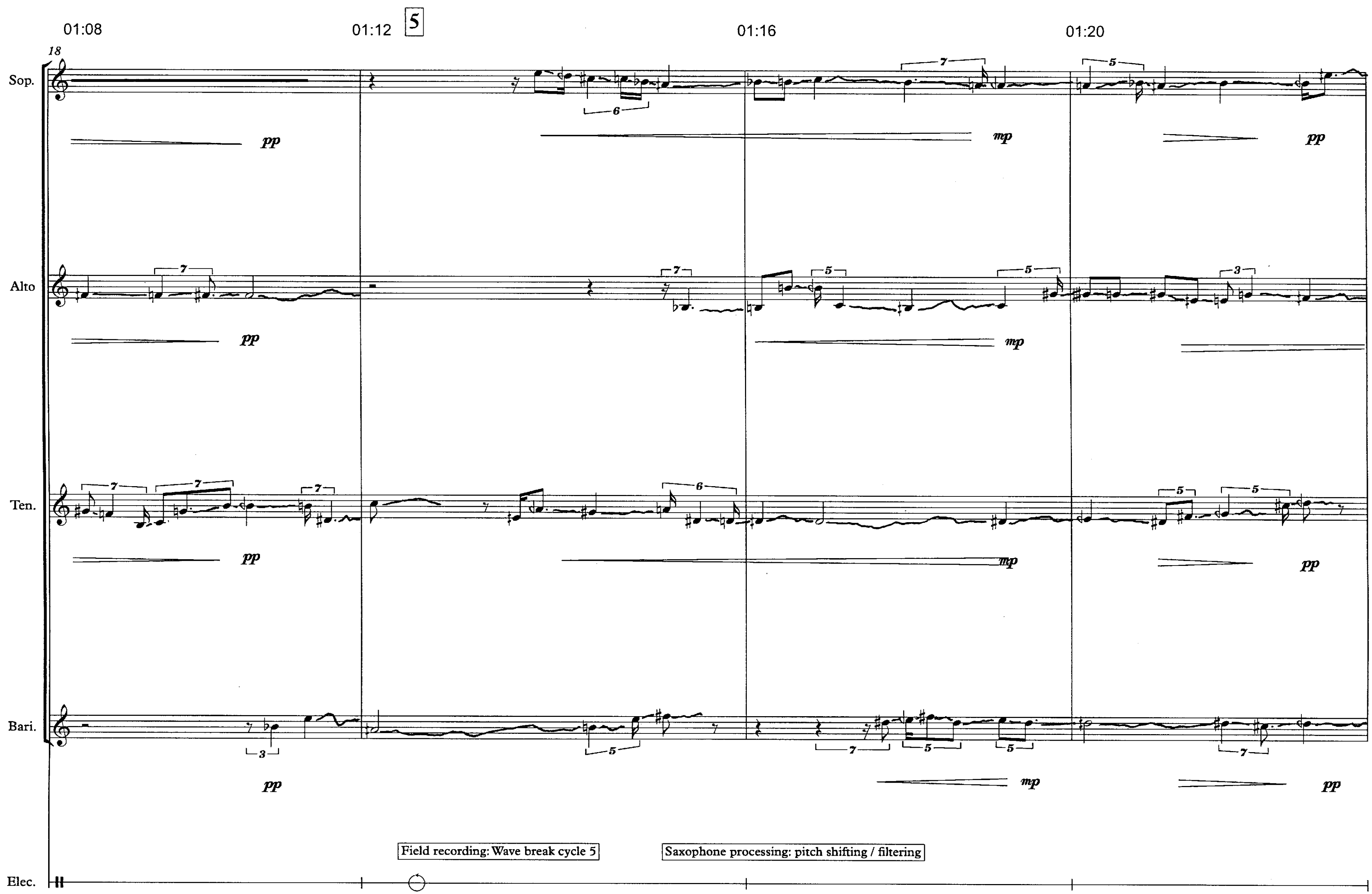
Christopher Luna-Mega (2017)

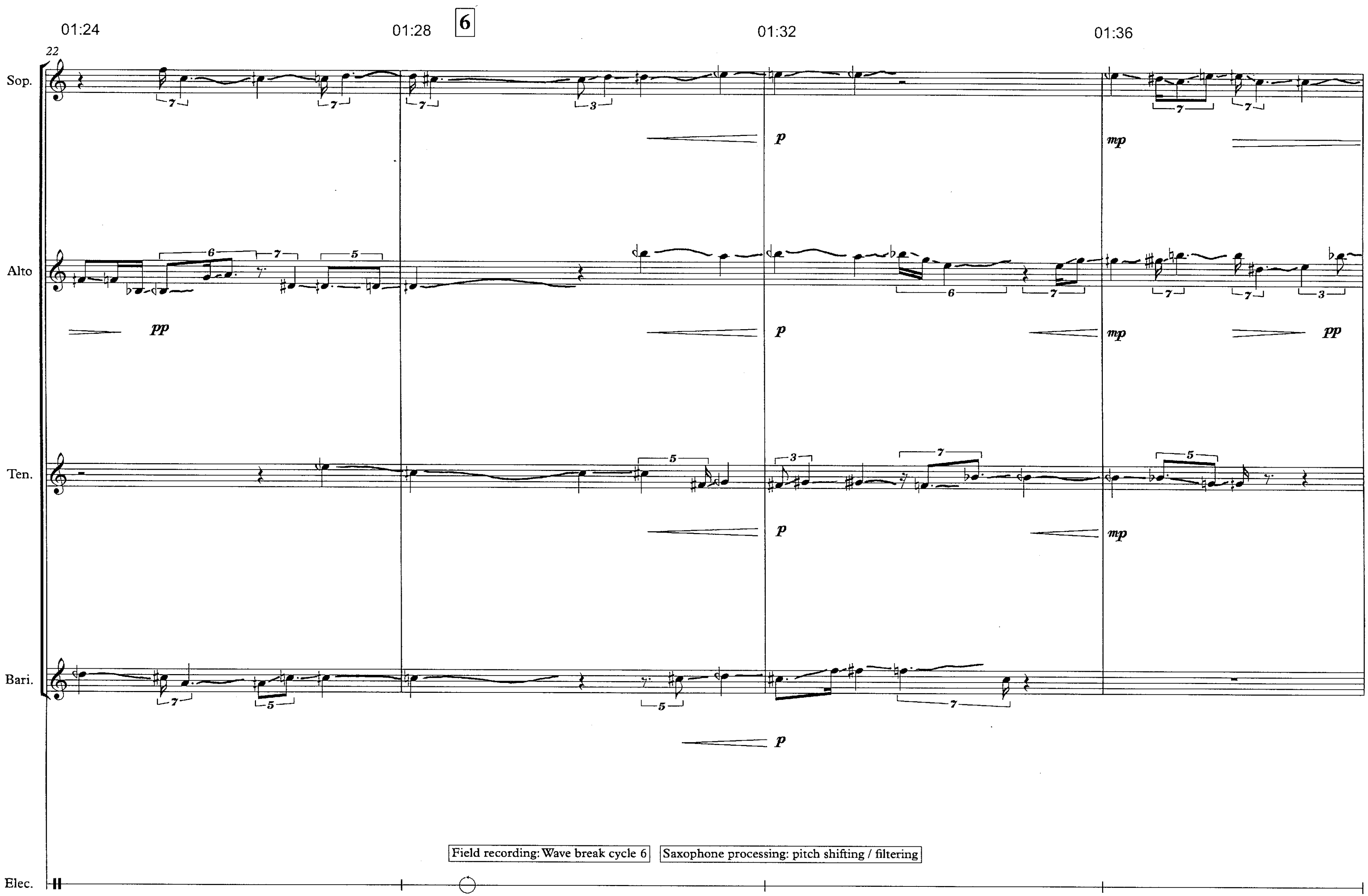


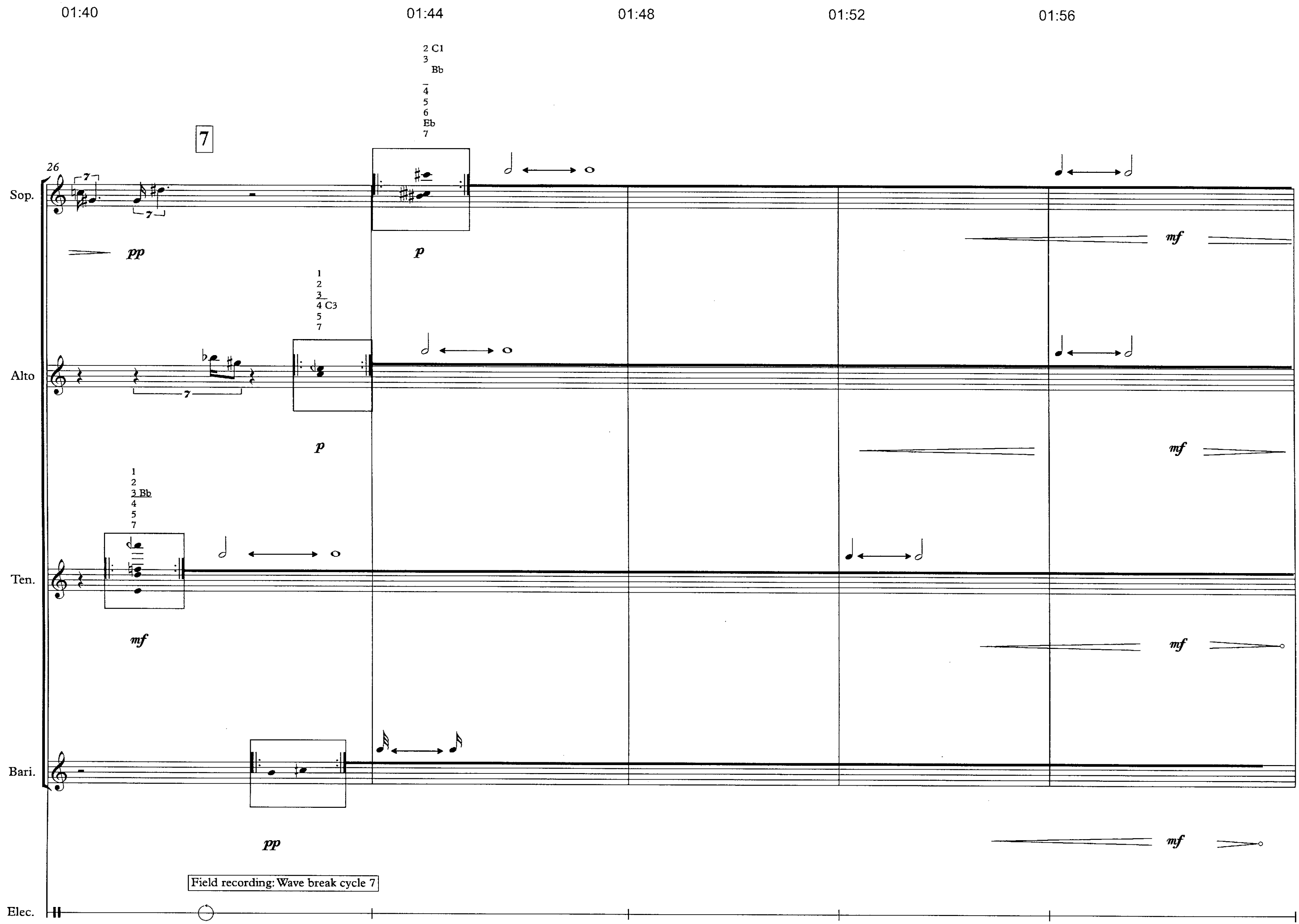

00:24
00:28
00:32
00:36
00:40
3
00:44
00:48
7
Sop.
Alto
Ten.
Bari.
f
p
f
Saxophone processing: amplification
Field recording: Wave break cycle 3
Saxophone processing: amplification
Elec.

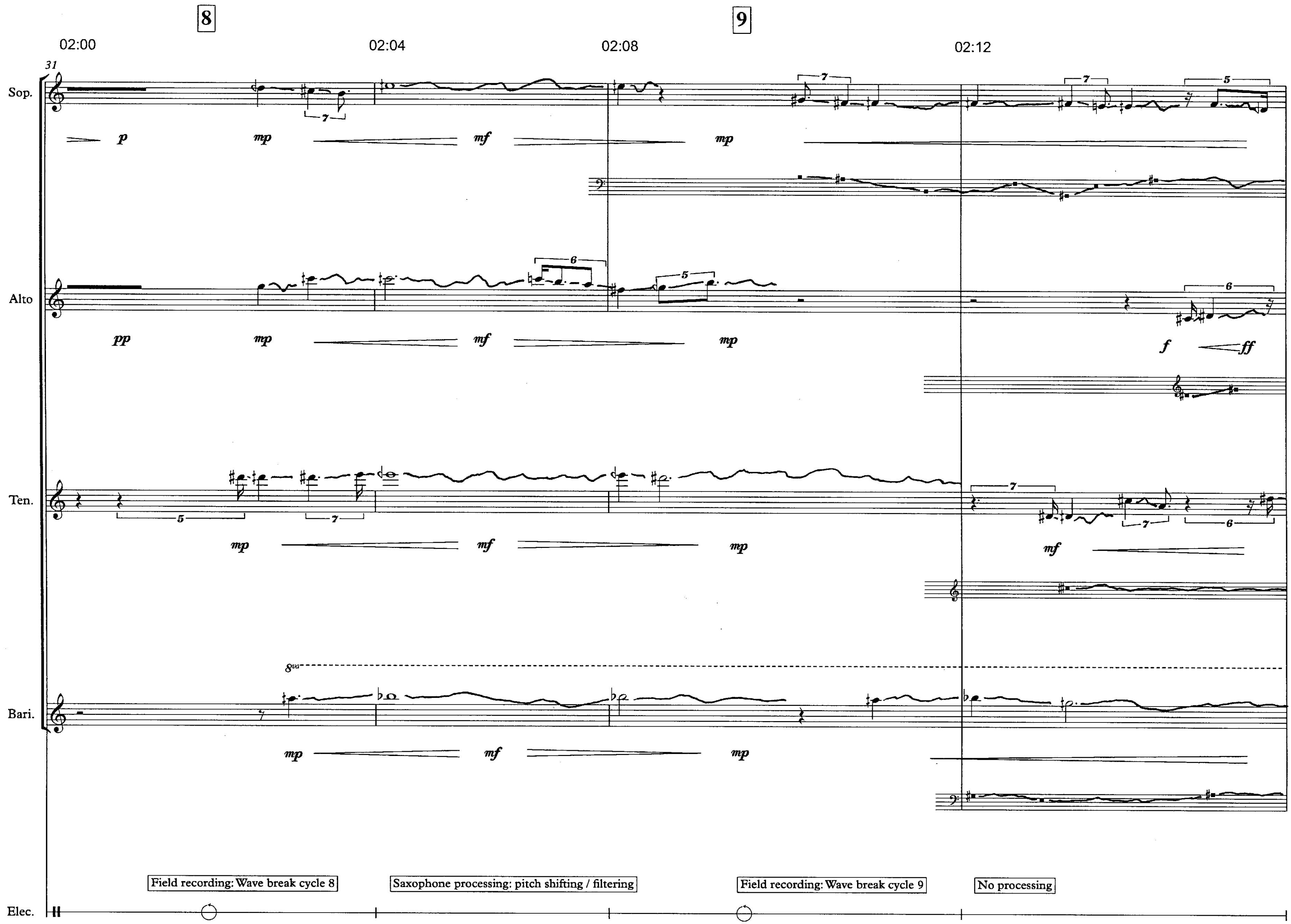


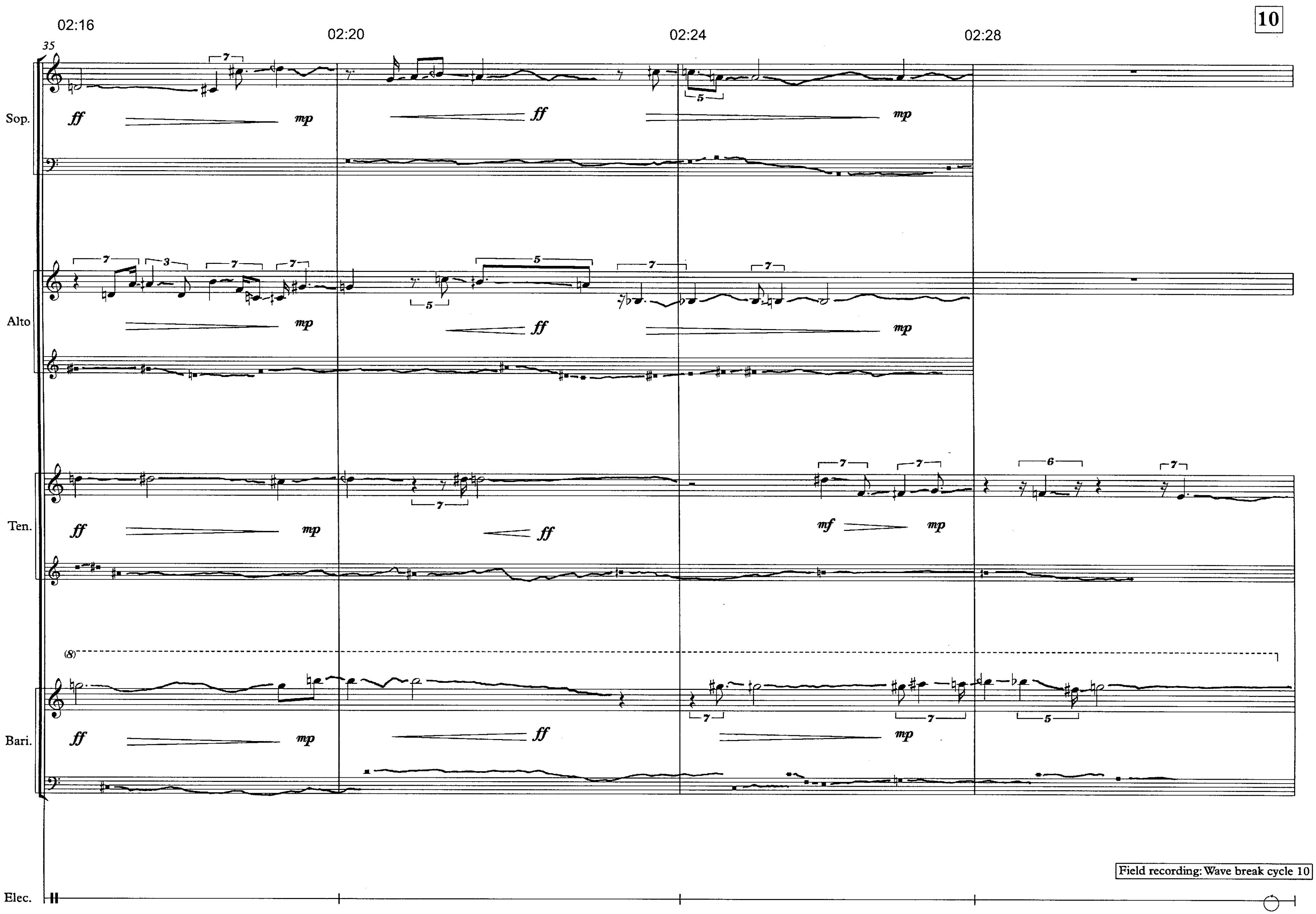
00:52
4
00:56
01:00
01:04
Sop.
Alto
Ten.
Bari.
Elec.
Field recording: Wave break cycle 4
Saxophone processing: pitch shifting / filtering

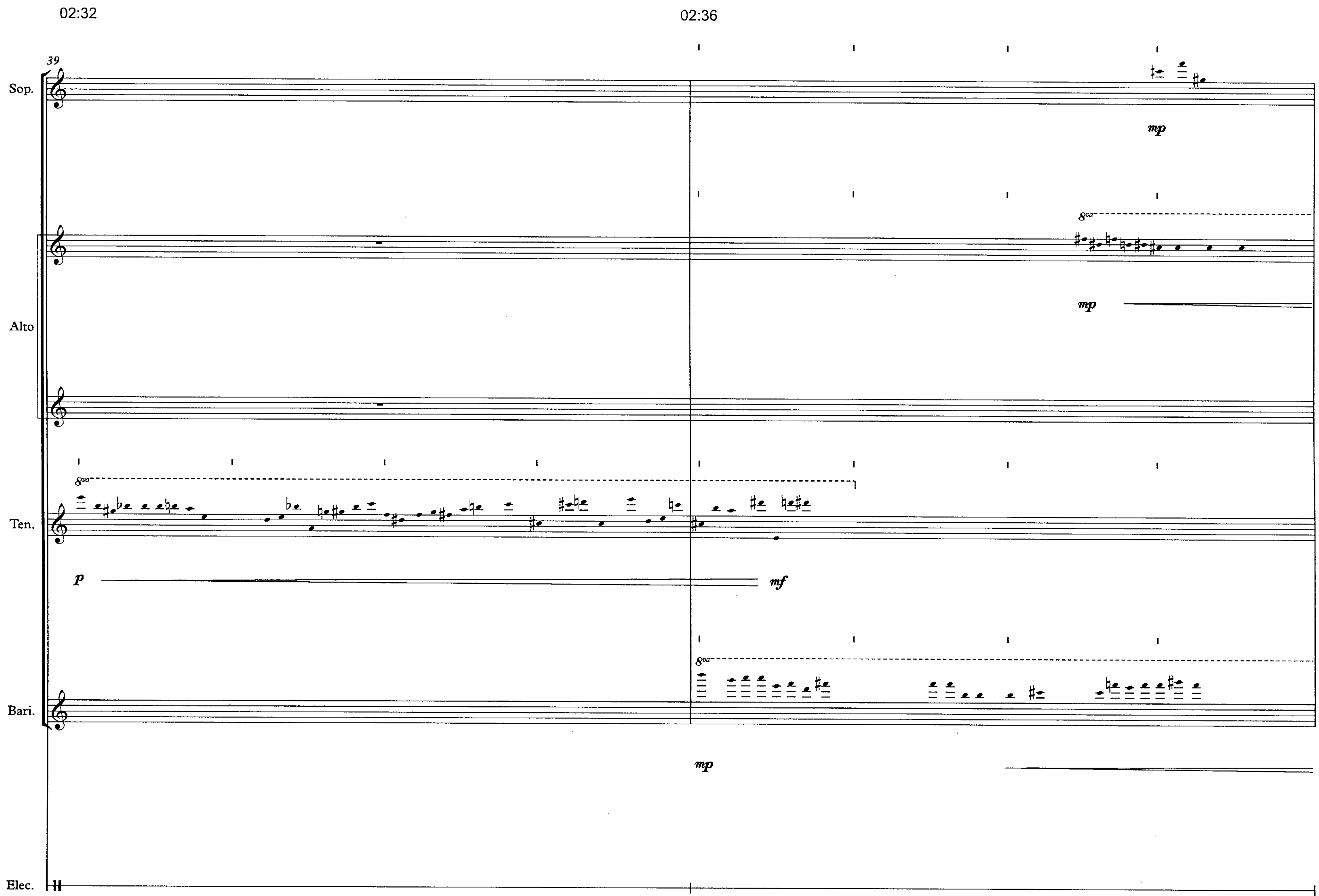

01:08
01:12
5
01:16
01:20
18
Sop.
Alto
Ten.
Bari.
Elec.
pp
mp
Field recording: Wave break cycle 5
Saxophone processing: pitch shifting / filtering

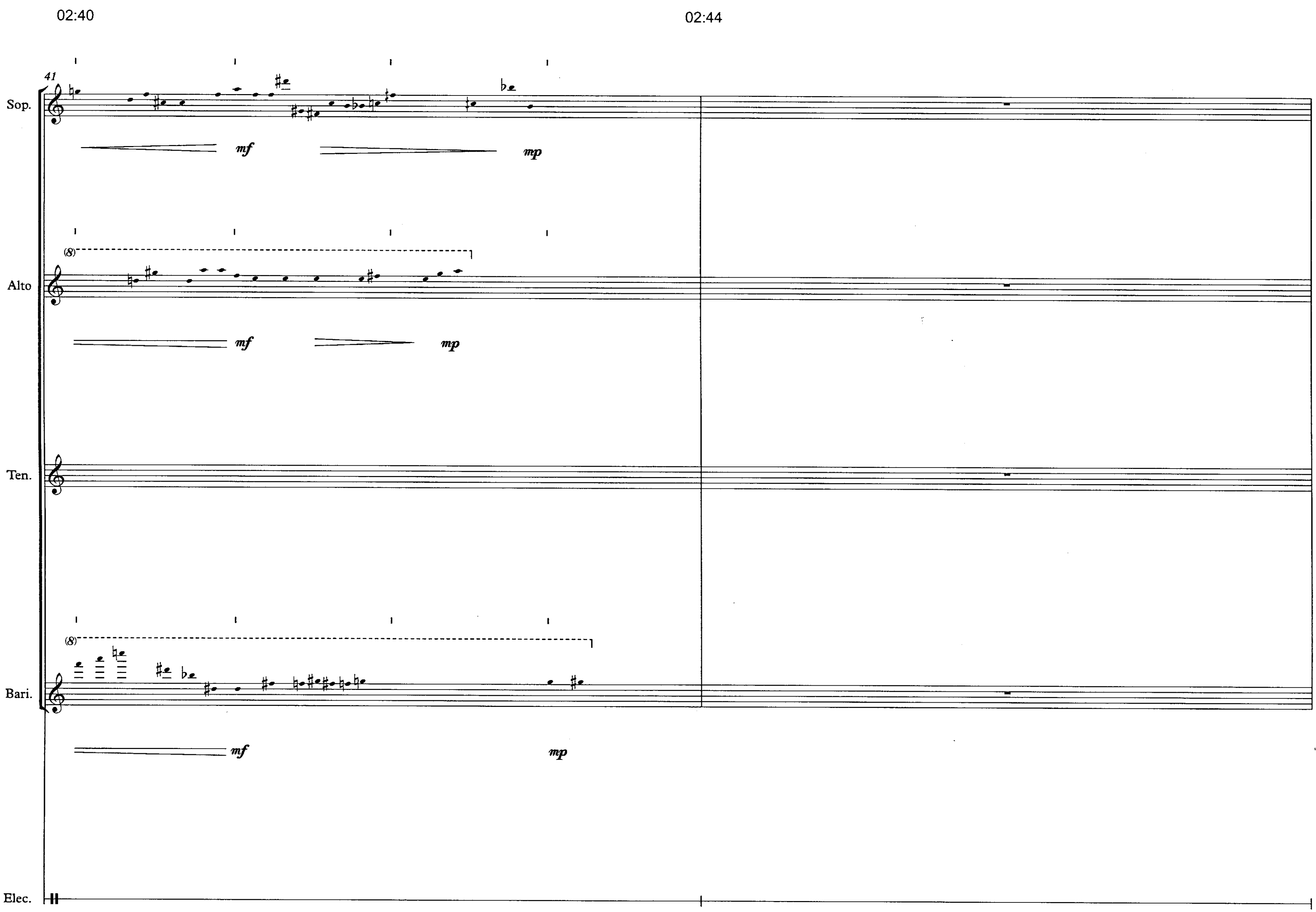

01:24
01:28
6
01:32
01:36
22
Sop.
Alto
Ten.
Bari.
Elec.
p
mp
pp
Field recording: Wave break cycle 6
Saxophone processing: pitch shifting / filtering

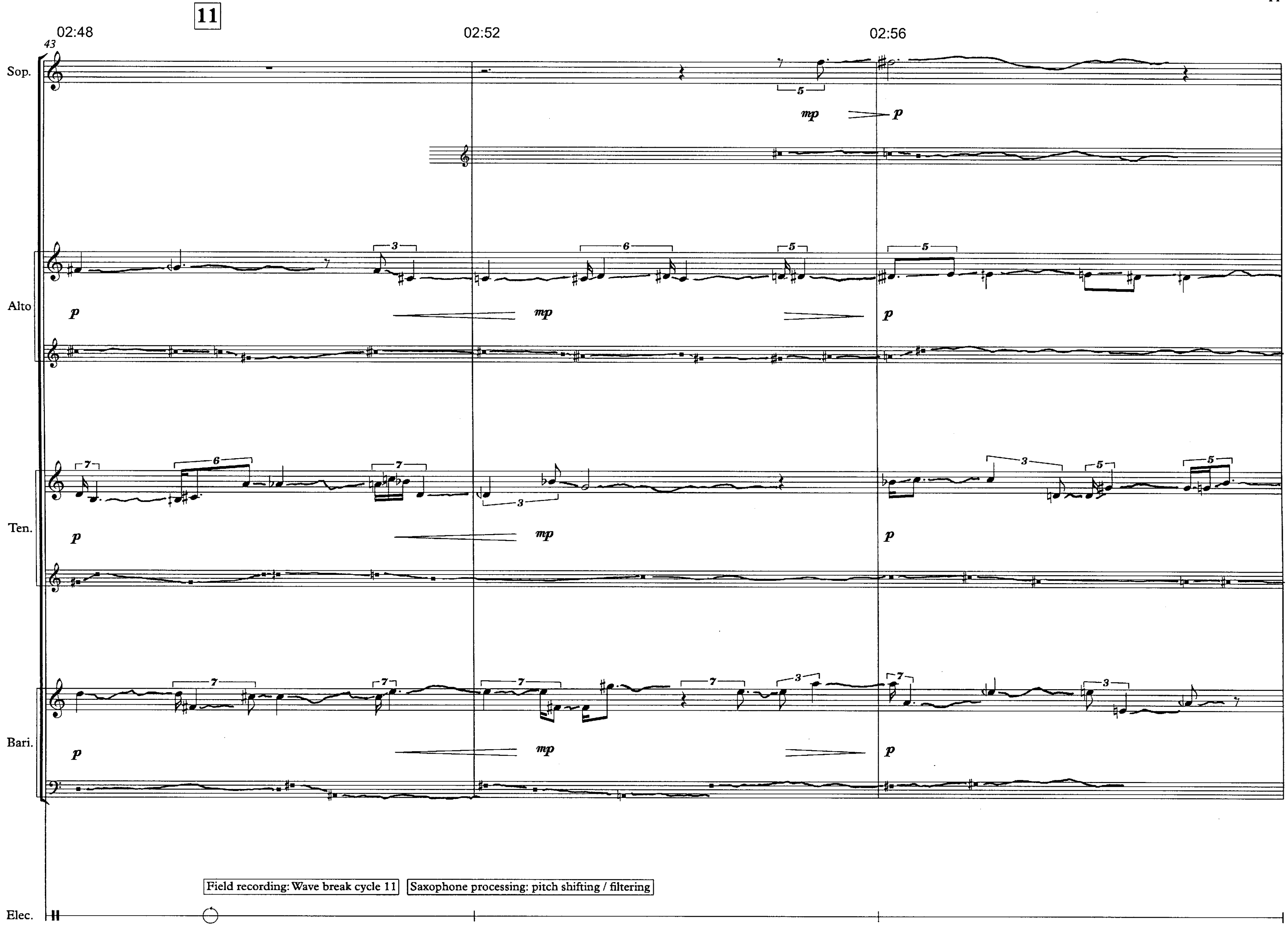

01:40
01:44
01:48
01:52
01:56
7
26
Sop.
Alto
Ten.
Bari.
Elec.
2 C1
3 Bb
4
5
6
Eb
7
1
2
3
4 C3
5
7
1
2
3 Bb
4
5
7
pp
p
mf
Field recording: Wave break cycle 7


8
9
02:00
02:04
02:08
02:12
31
Sop.
Alto
Ten.
Bari.
Elec.
8va
Field recording: Wave break cycle 8
Saxophone processing: pitch shifting / filtering
Field recording: Wave break cycle 9
No processing

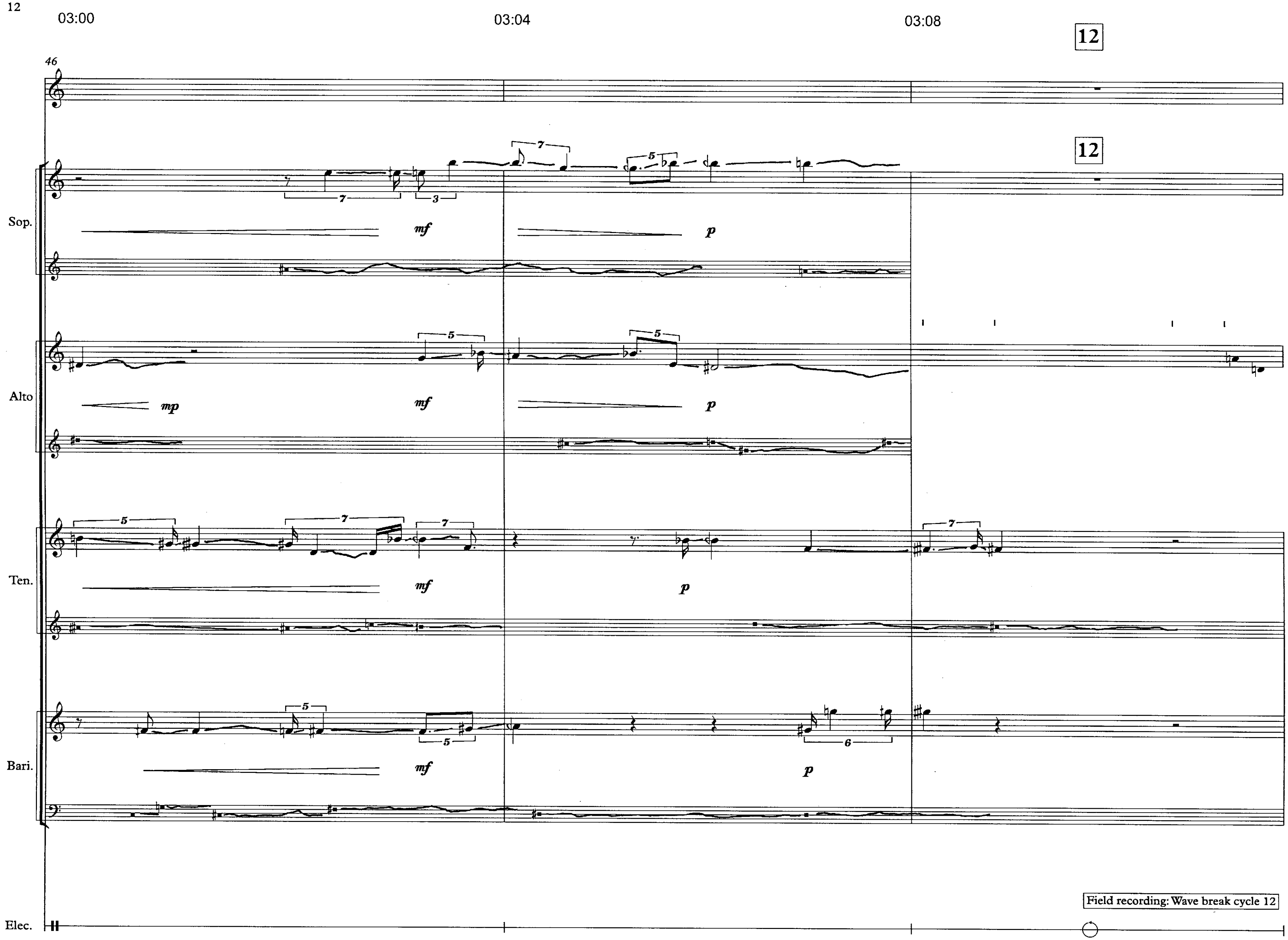

02:16
02:20
02:24
02:28
10
35
Sop.
ff
mp
ff
mp
Alto
mp
ff
mp
Ten.
ff
mp
ff
mf
mp
(8)
Bari.
ff
mp
ff
mp
Field recording: Wave break cycle 10
Elec.

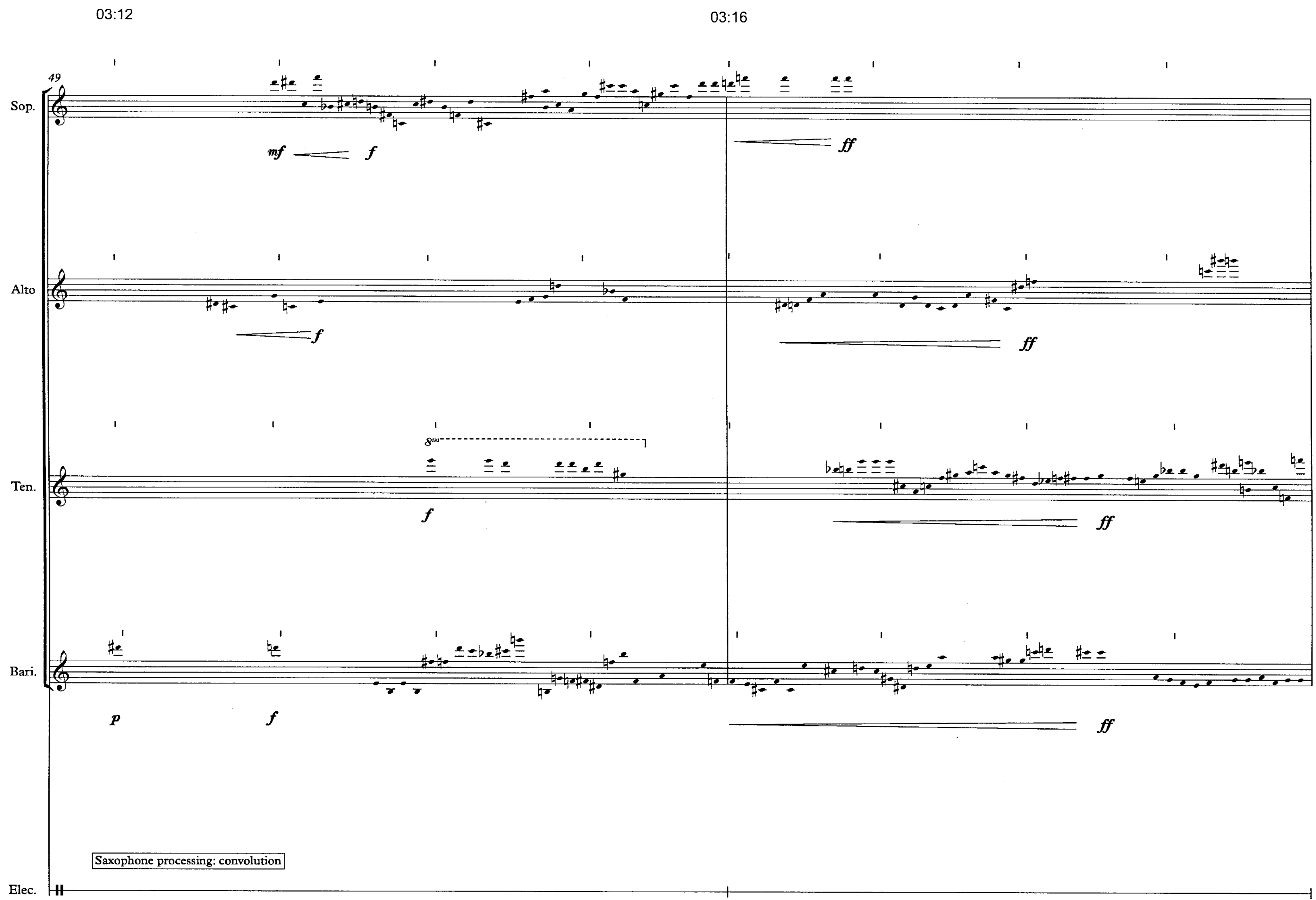

02:32
02:36
39
Sop.
mp
Alto
8va
mp
Ten.
8va
p
mf
Bari.
8va
mp
Elec.

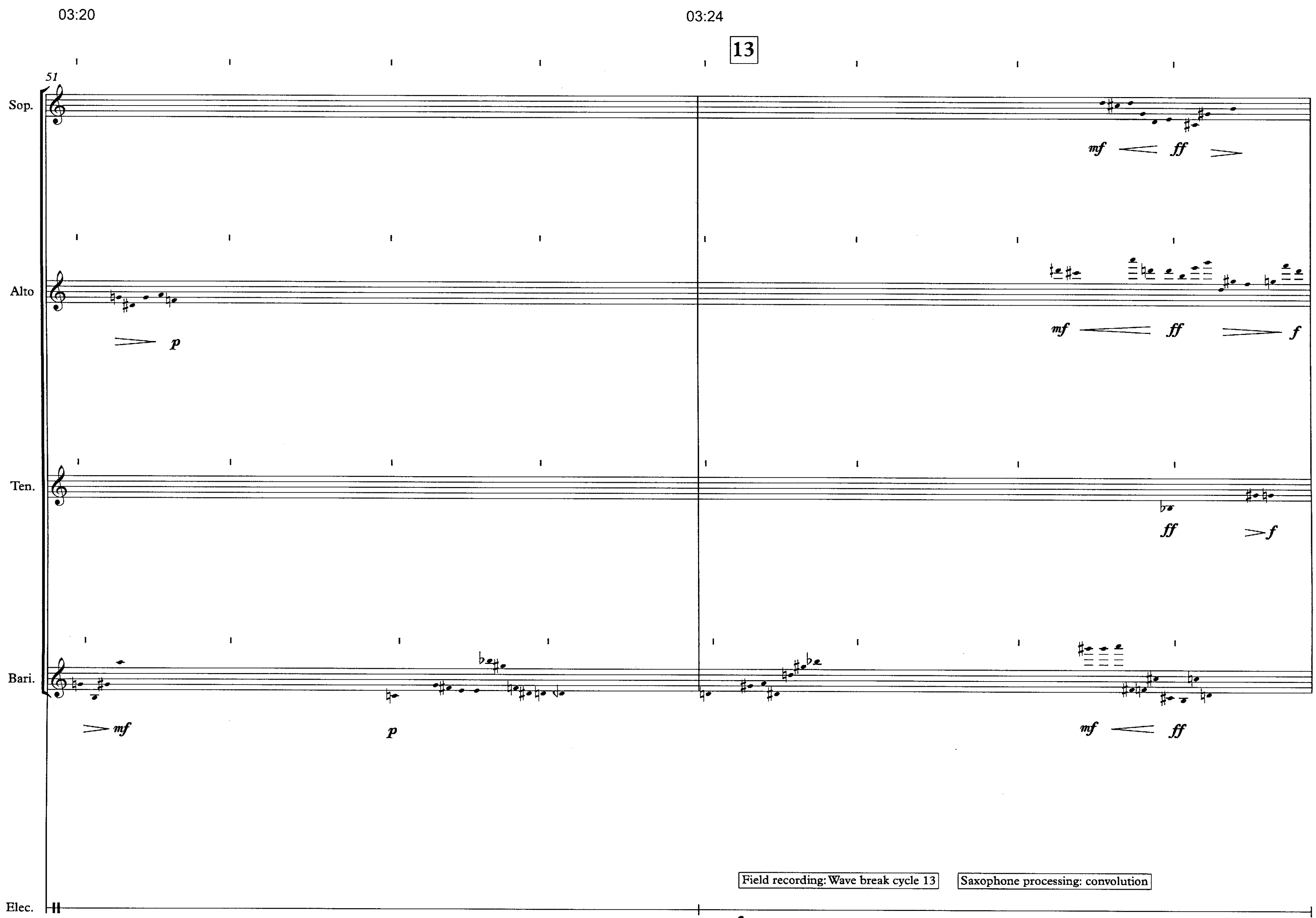

02:40
02:44
41
Sop.
mf
mp
(8)
Alto
mf
mp
Ten.
(8)
Bari.
mf
mp
Elec.

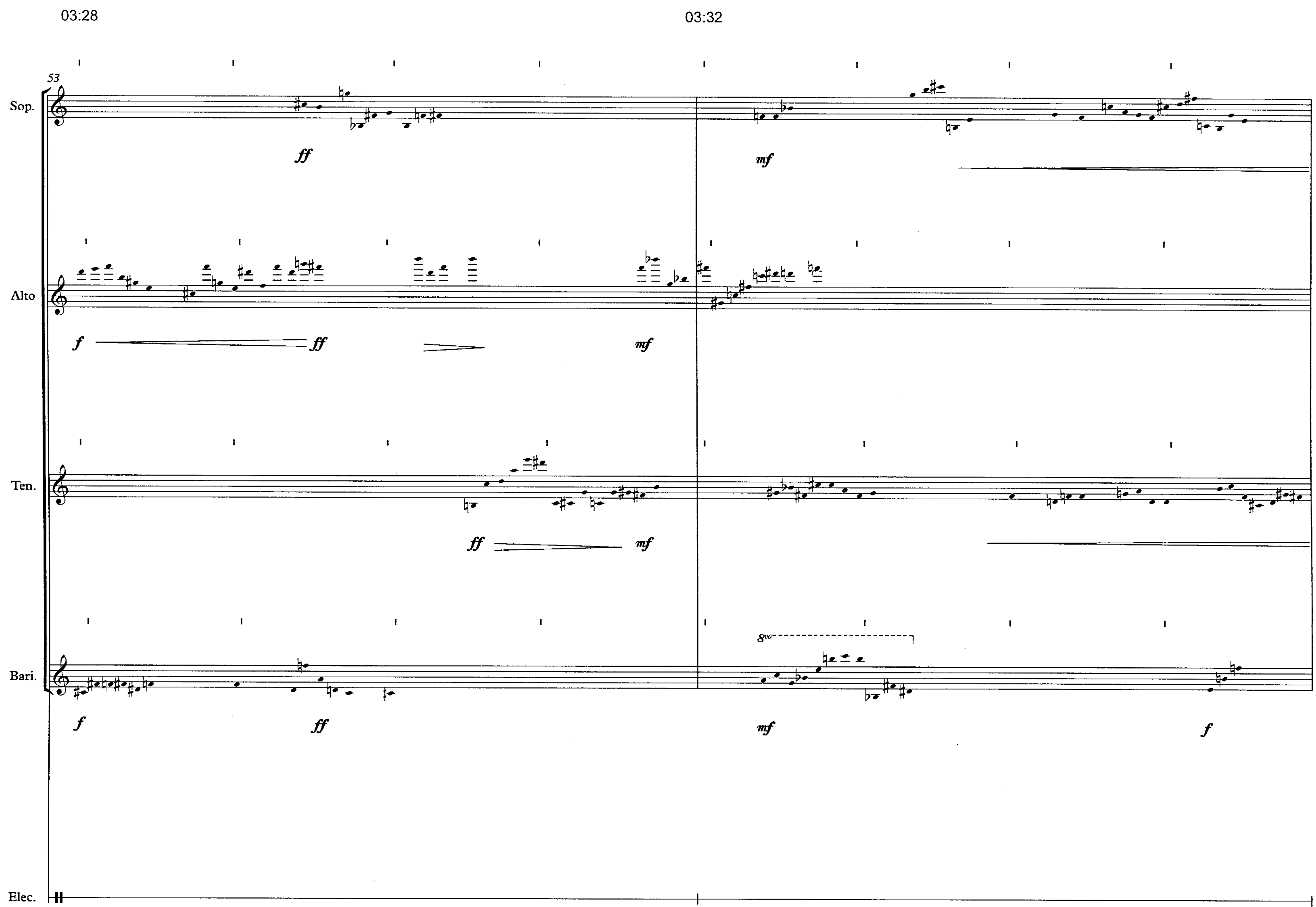


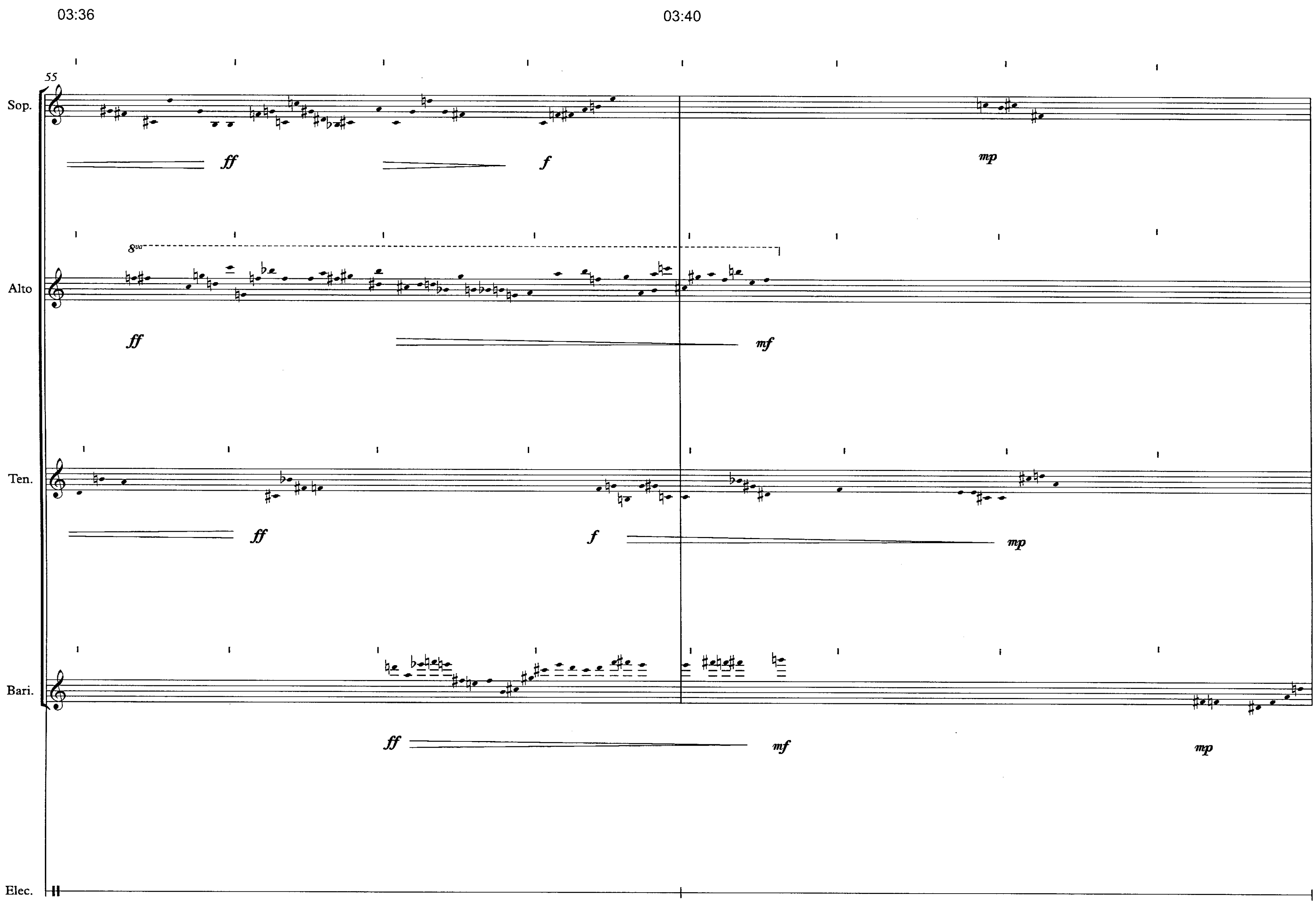
11
02:48
02:52
02:56
43
Sop.
Alto
Ten.
Bari.
Elec.
Field recording: Wave break cycle 11
Saxophone processing: pitch shifting / filtering

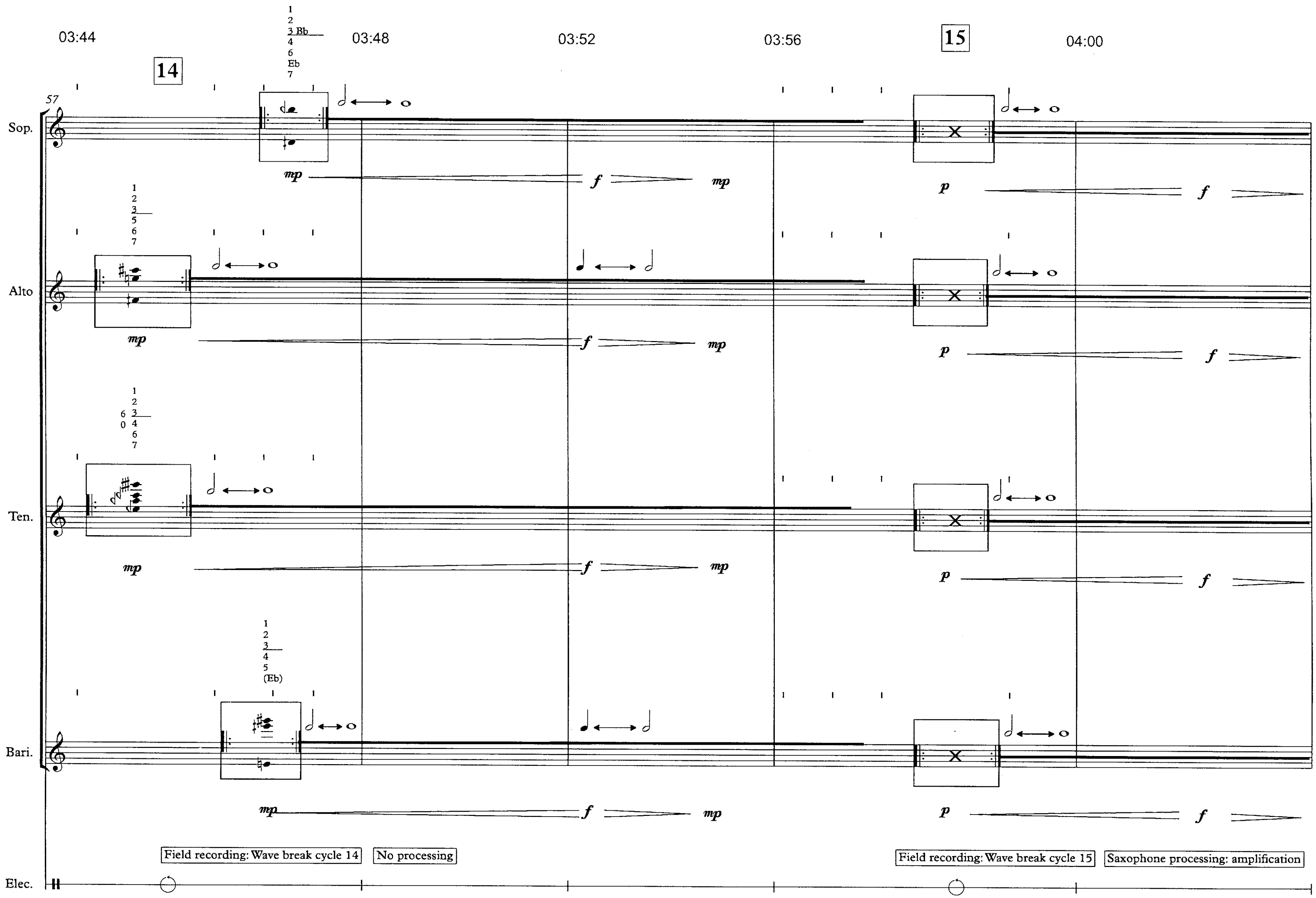

03:00
03:04
03:08
12
46
12
Sop.
mf
p
Alto
mp
mf
p
Ten.
mf
p
Bari.
mf
p
Field recording: Wave break cycle 12
Elec.

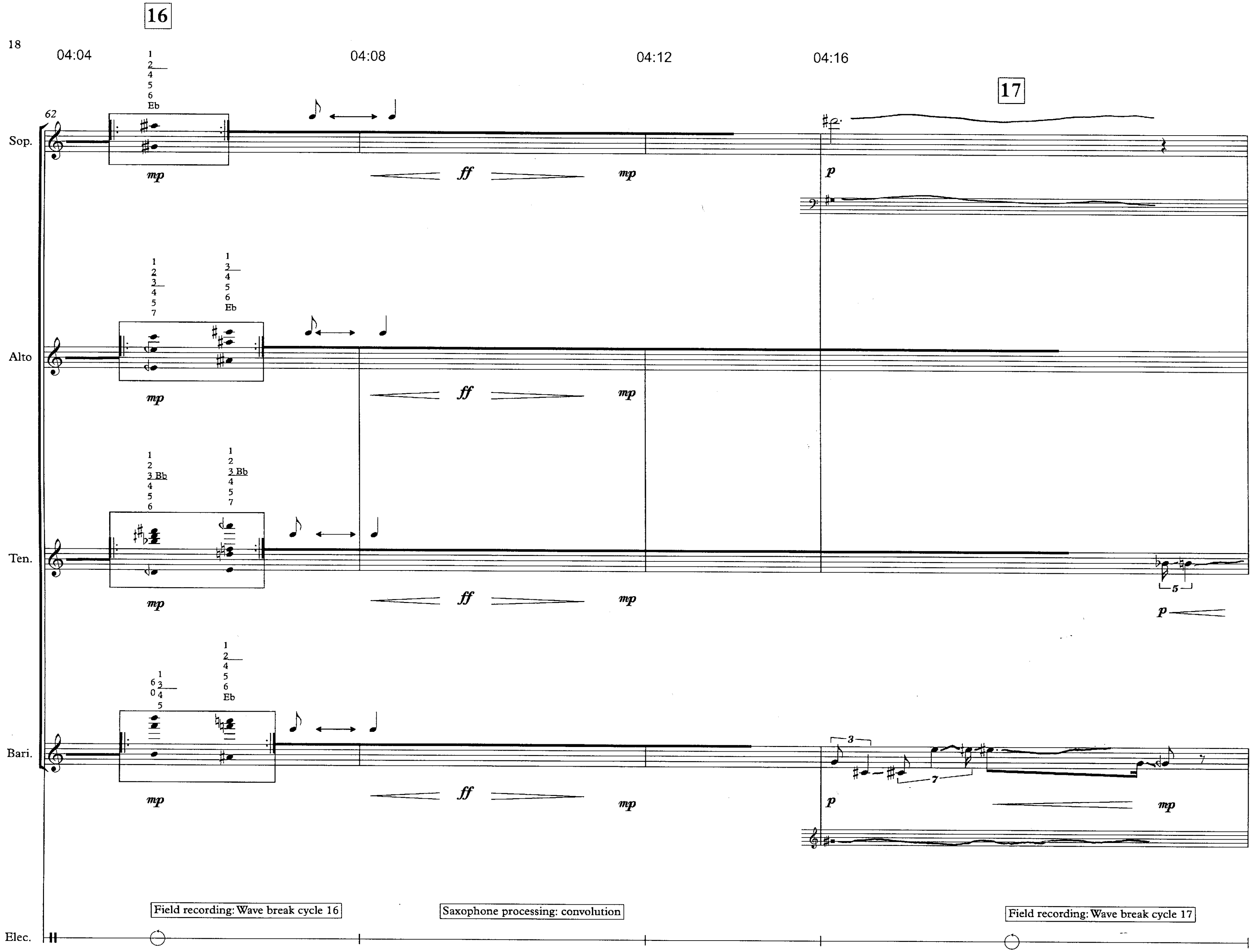

03:12
03:16
49
Sop.
mf
f
ff
Alto
f
ff
Ten.
8va
f
ff
Bari.
p
f
ff
Saxophone processing: convolution
Elec.

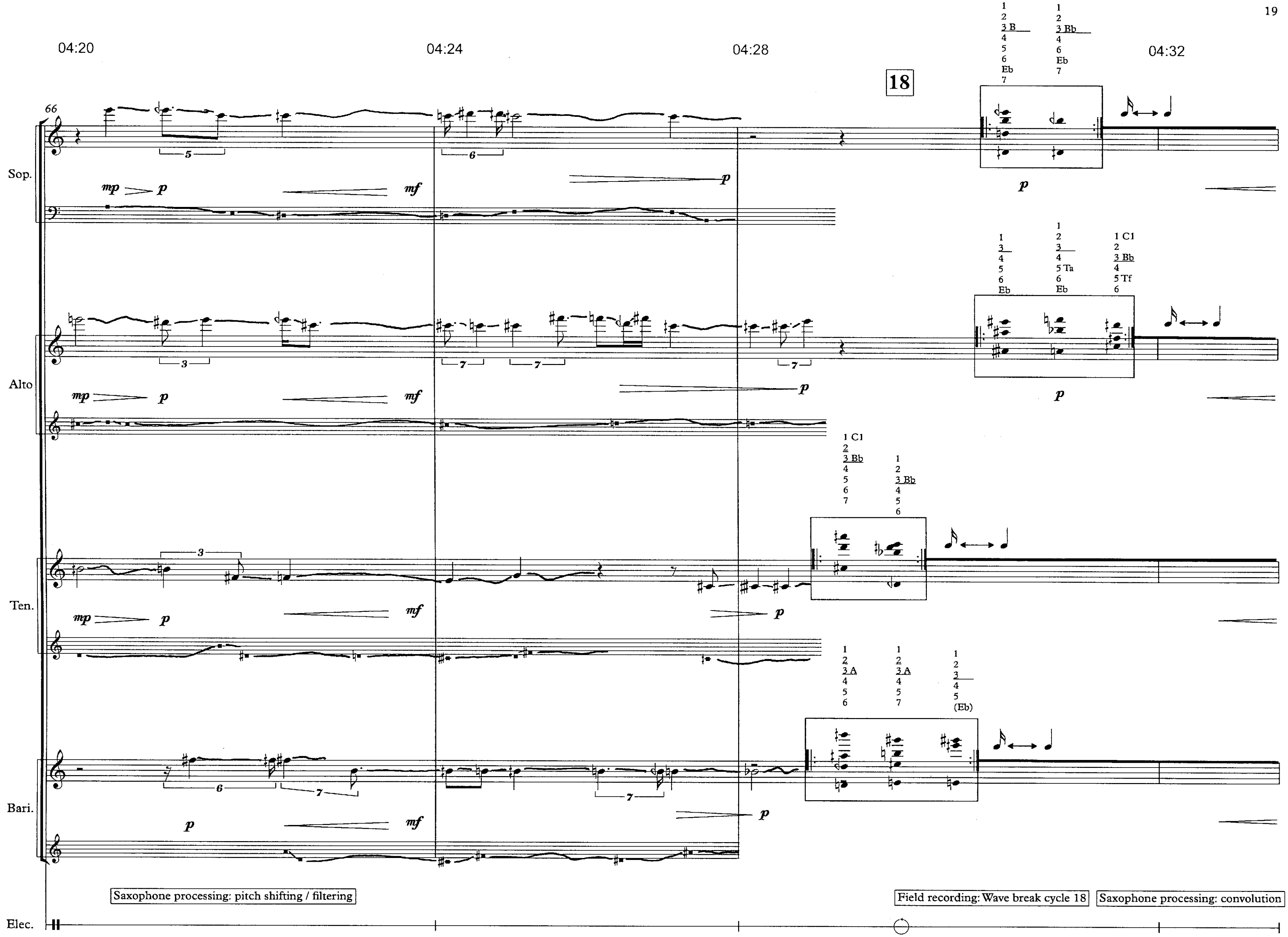

03:20
03:24
13
51
Sop.
Alto
Ten.
Bari.
Elec.
mf ff
p
mf ff f
ff f
mf
p
mf ff
Field recording: Wave break cycle 13
Saxophone processing: convolution

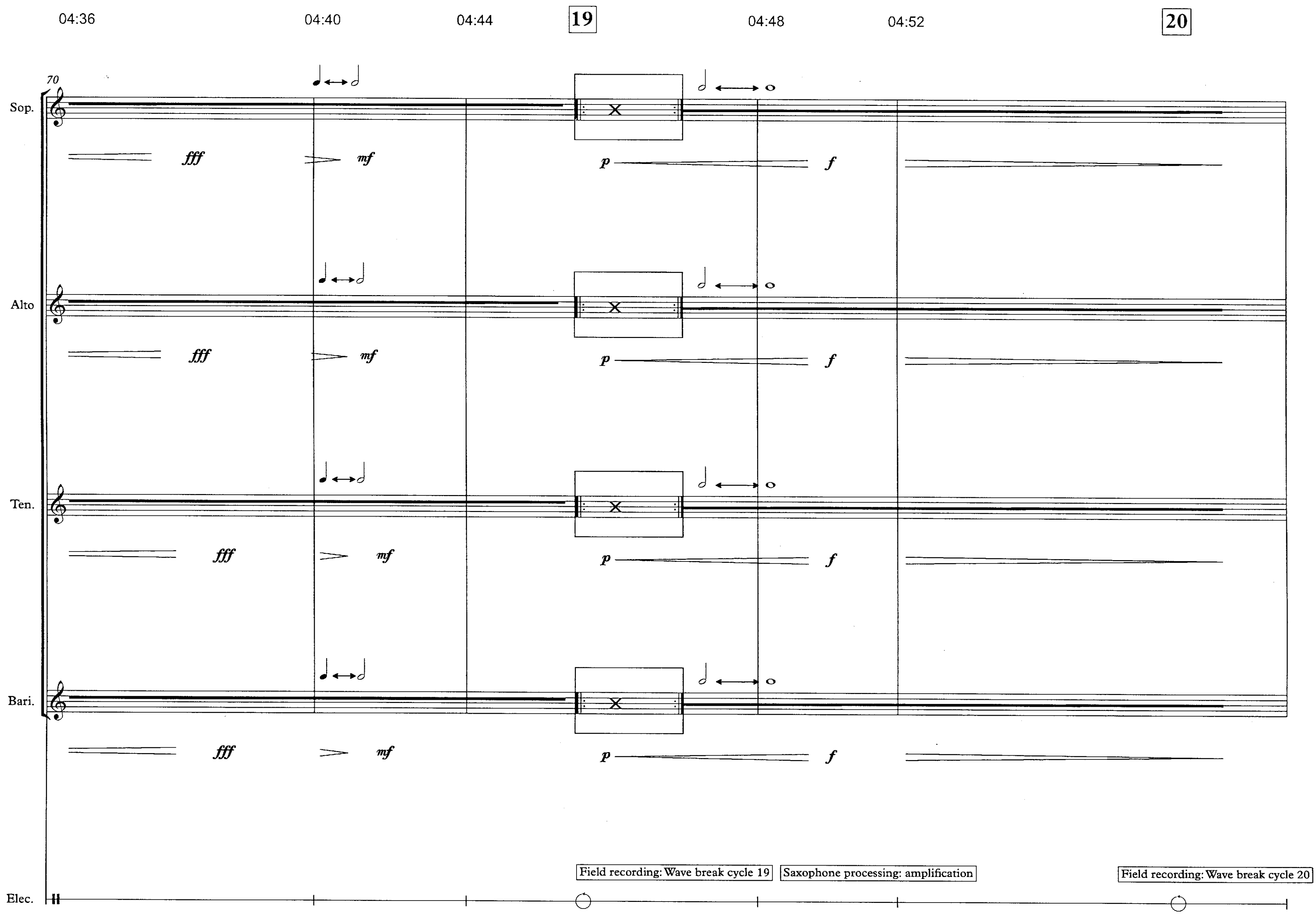

03:28
03:32
53
Sop.
ff
mf
Alto
f
ff
mf
Ten.
ff
mf
Bari.
8va
f
ff
mf
f
Elec.


03:36
03:40
55
Sop.
ff
f
mp
Alto
8va
ff
mf
Ten.
ff
f
mp
Bari.
ff
mf
mp
Elec.

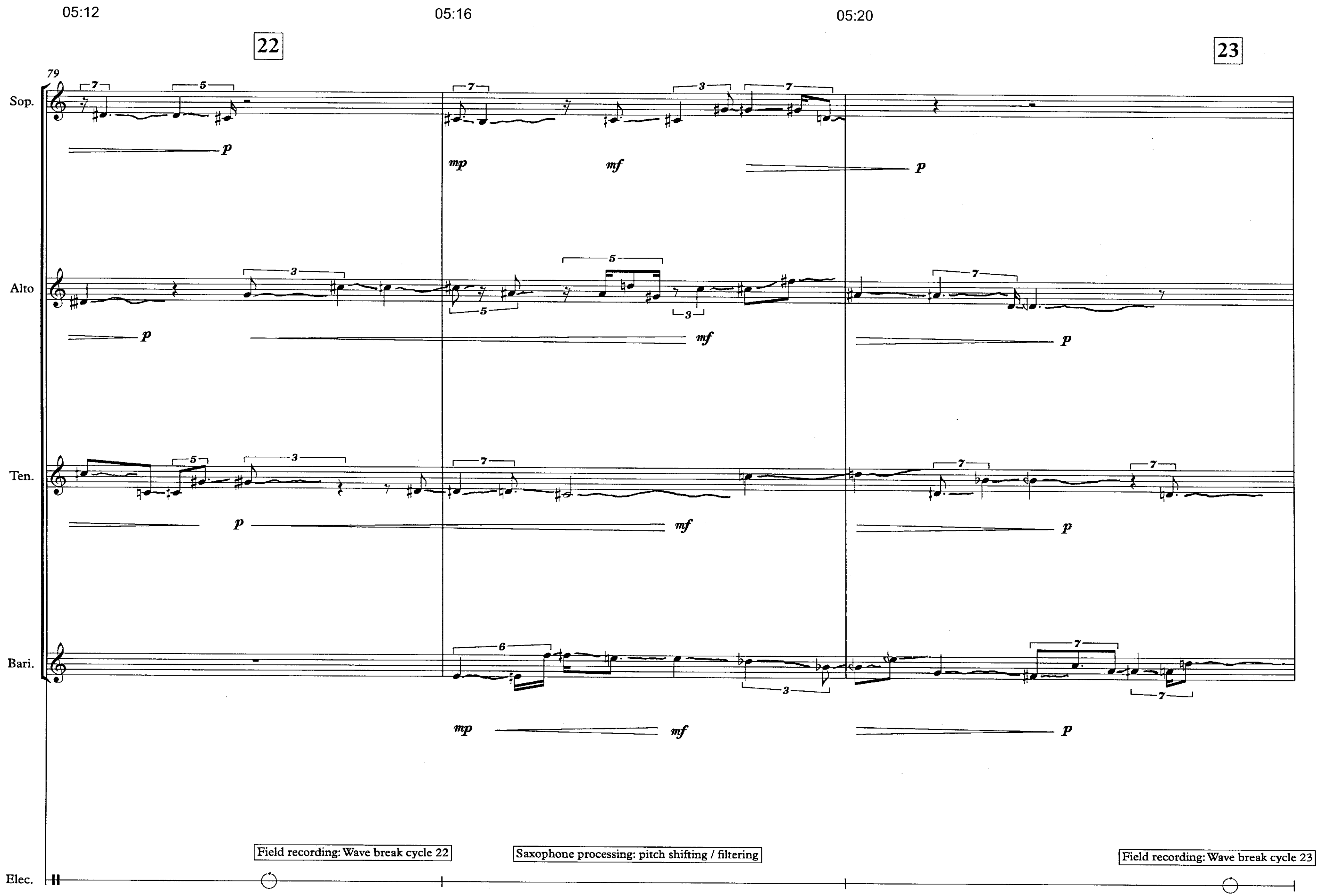

03:44
14
03:48
03:52
03:56
15
04:00
57
Sop.
Alto
Ten.
Bari.
Elec.
mp
f
p
Field recording: Wave break cycle 14
No processing
Field recording: Wave break cycle 15
Saxophone processing: amplification

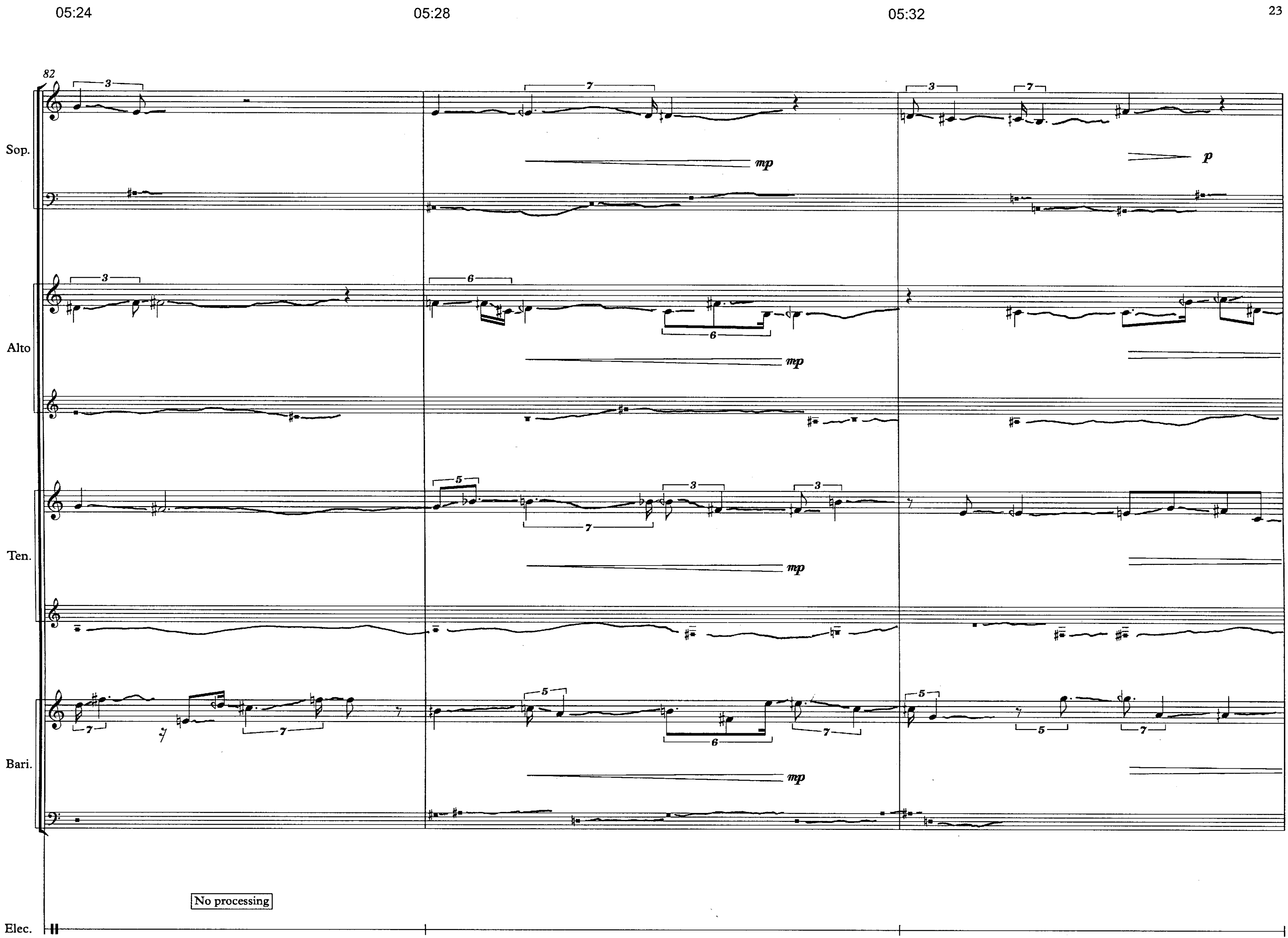

16
04:04
04:08
04:12
04:16
17
62
Sop.
Alto
Ten.
Bari.
Elec.
mp
ff
p
Field recording: Wave break cycle 16
Saxophone processing: convolution
Field recording: Wave break cycle 17

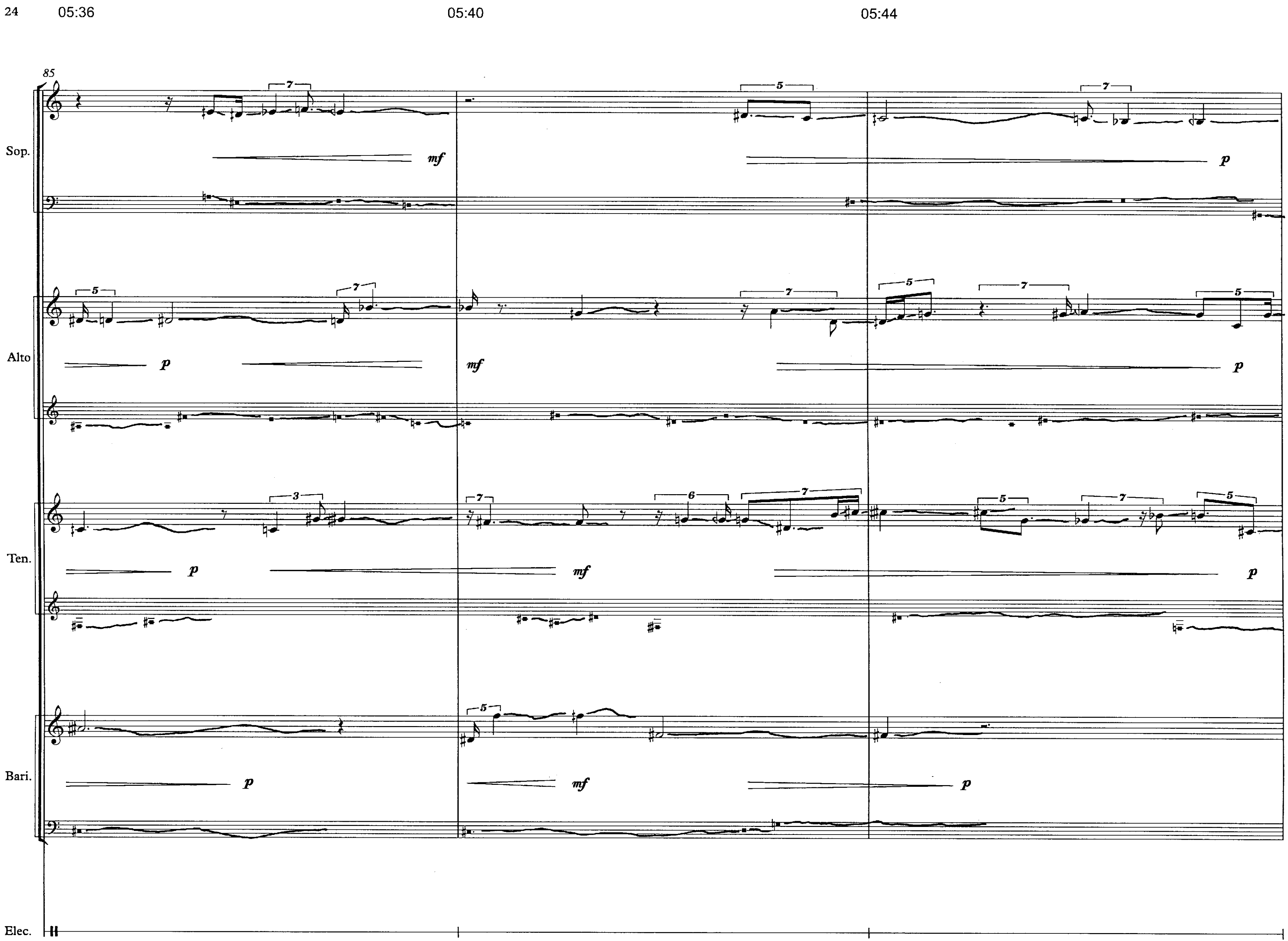

04:20
04:24
04:28
04:32
18
66
Sop.
Alto
Ten.
Bari.
Elec.
mp p mf p
Saxophone processing: pitch shifting / filtering
Field recording: Wave break cycle 18
Saxophone processing: convolution

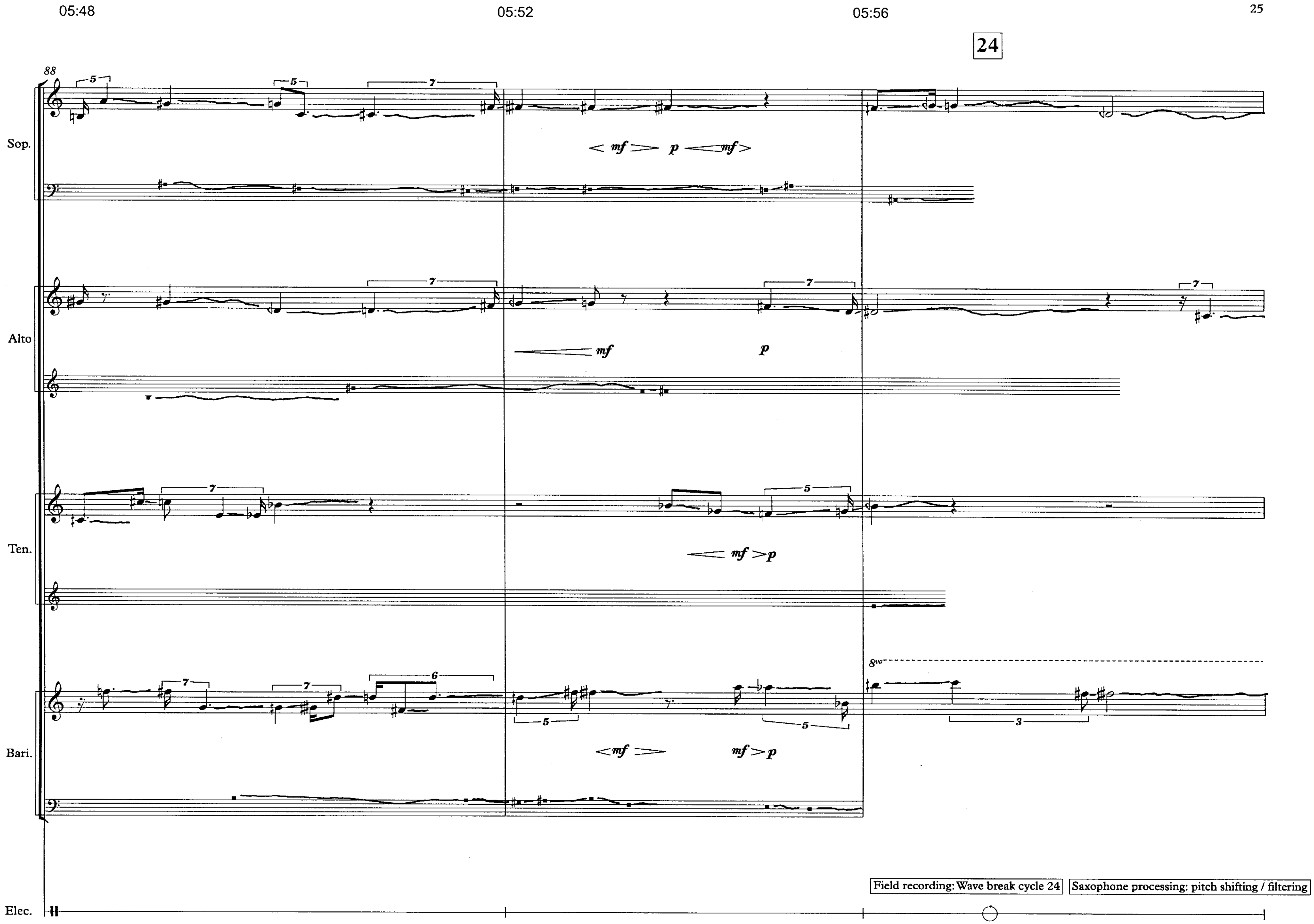

04:36
04:40
04:44
19
04:48
04:52
20
70
Sop.
fff
mf
p
f
Alto
fff
mf
p
f
Ten.
fff
mf
p
f
Bari.
fff
mf
p
f
Field recording: Wave break cycle 19
Saxophone processing: amplification
Field recording: Wave break cycle 20
Elec.


04:56
05:00
21
05:04
05:08
75
Sop.
Alto
Ten.
Bari.
Elec.
No processing
Field recording: Wave break cycle 21
No processing

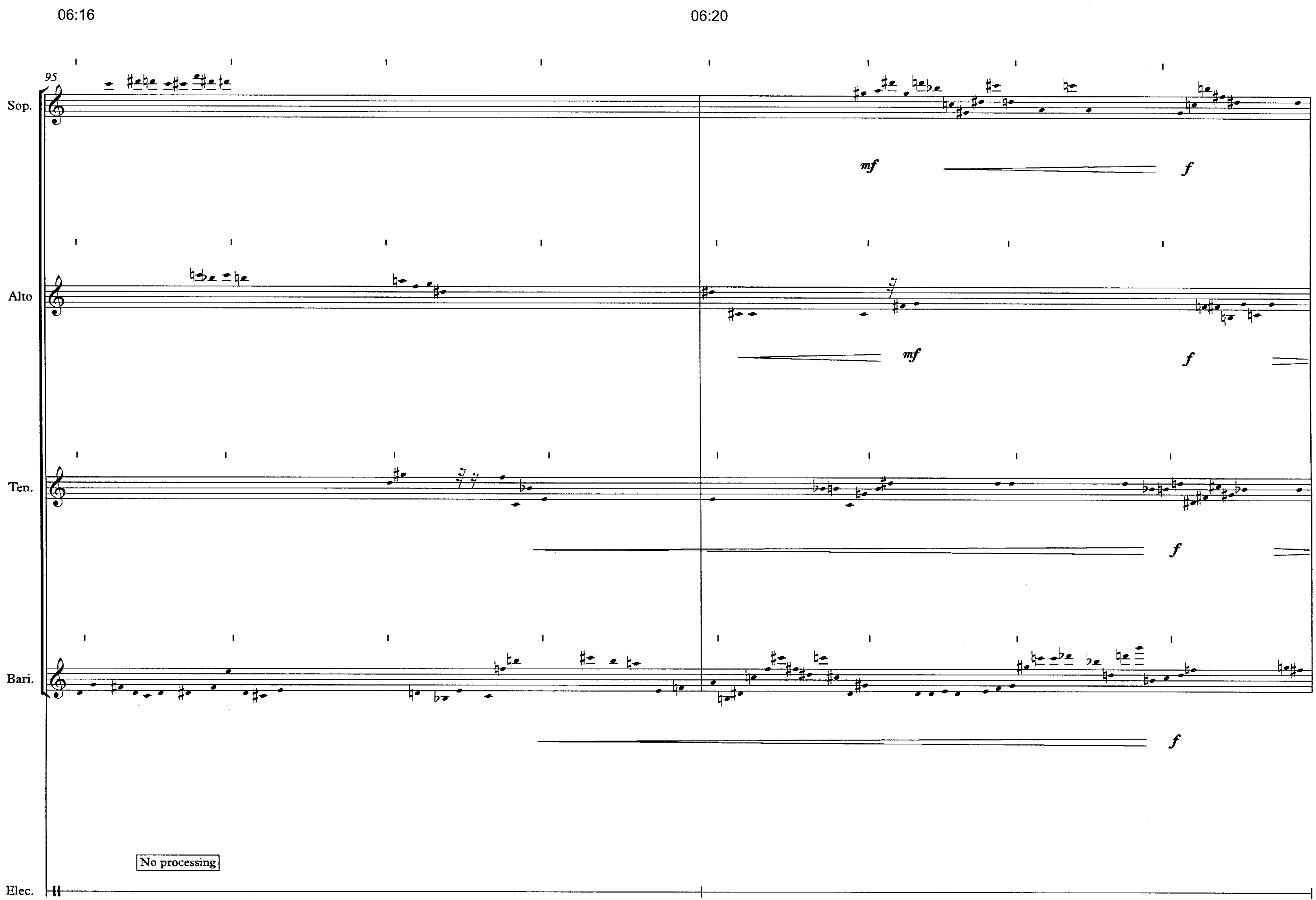

05:12
05:16
05:20
22
23
79
Sop.
Alto
Ten.
Bari.
Elec.
Field recording: Wave break cycle 22
Saxophone processing: pitch shifting / filtering
Field recording: Wave break cycle 23

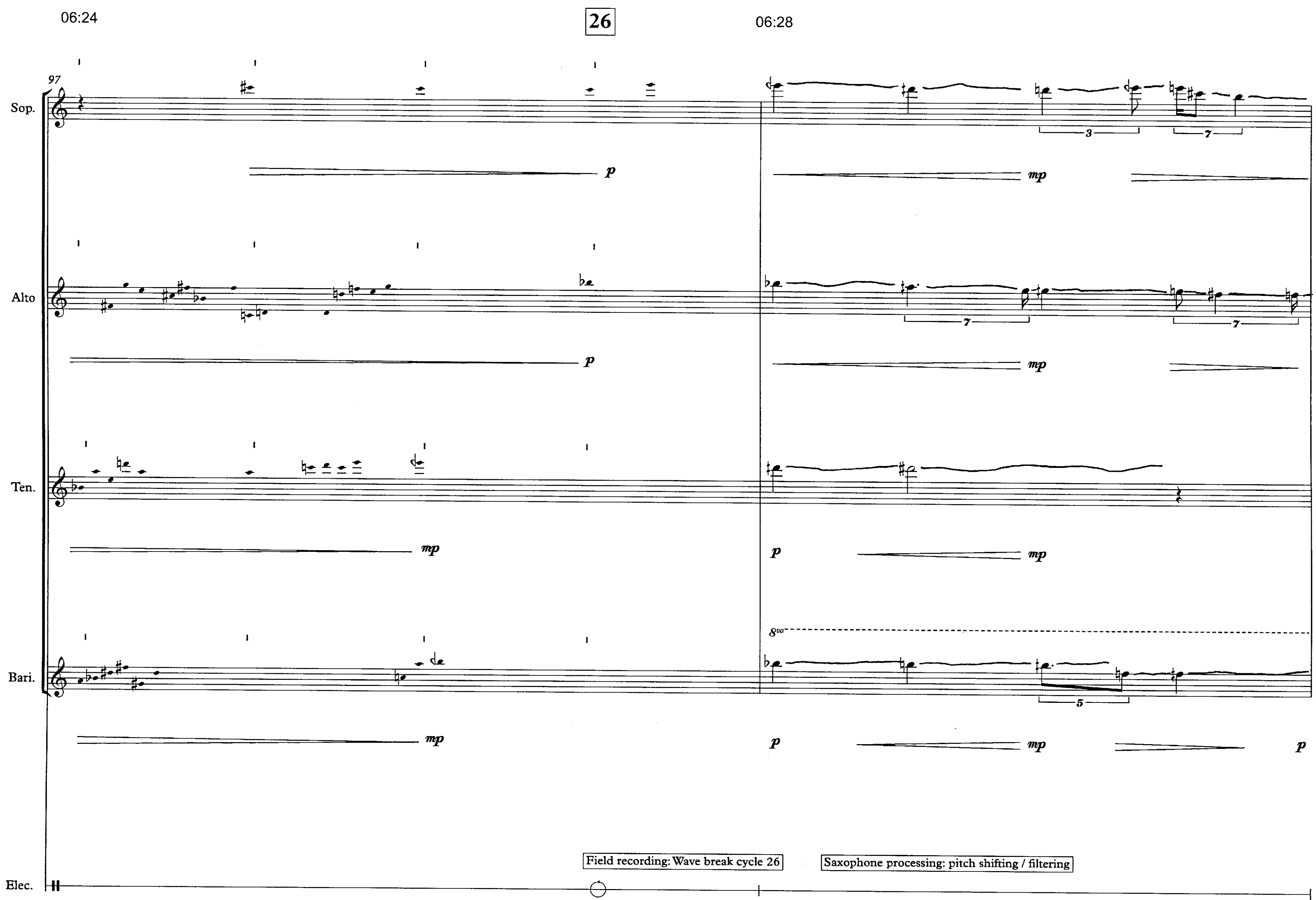

05:24
05:28
05:32
82
Sop.
Alto
Ten.
Bari.
Elec.
No processing
mp
p

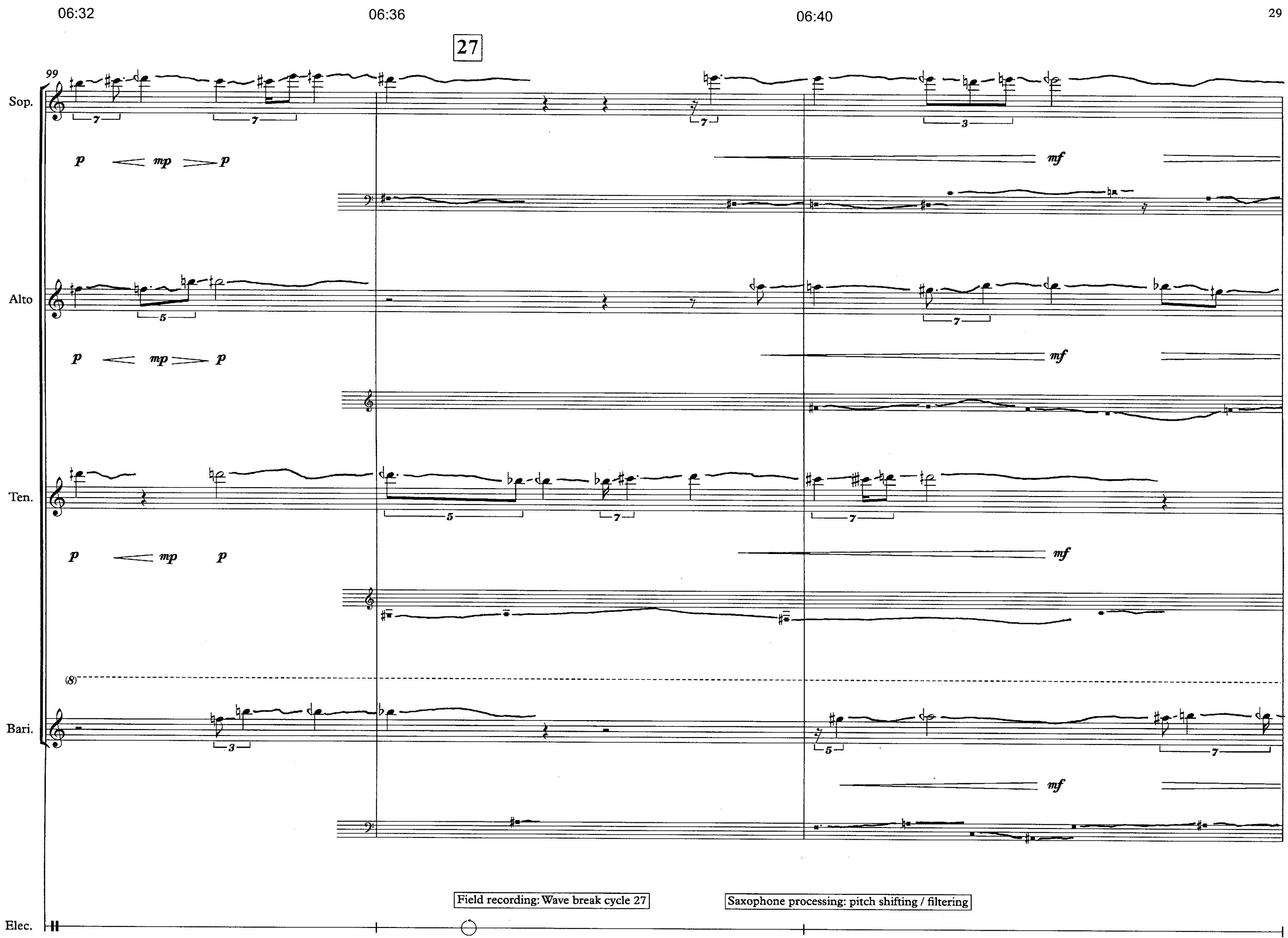

05:36
05:40
05:44
85
Sop.
mf
p
Alto
p
mf
p
Ten.
p
mf
p
Bari.
p
mf
p
Elec.

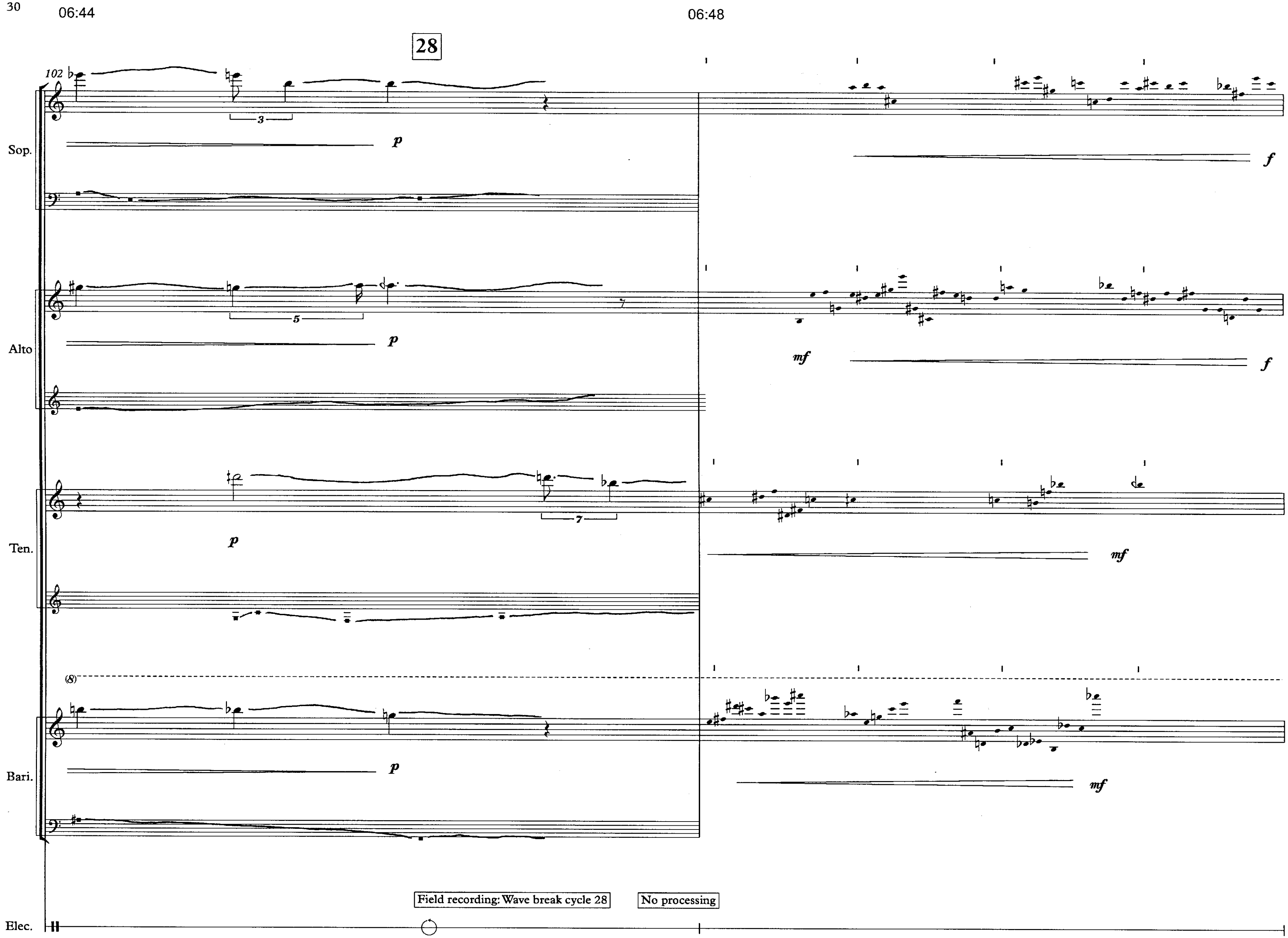

05:48
05:52
05:56
24
88
Sop.
Alto
Ten.
Bari.
Elec.
mf
p
8va
Field recording: Wave break cycle 24
Saxophone processing: pitch shifting / filtering

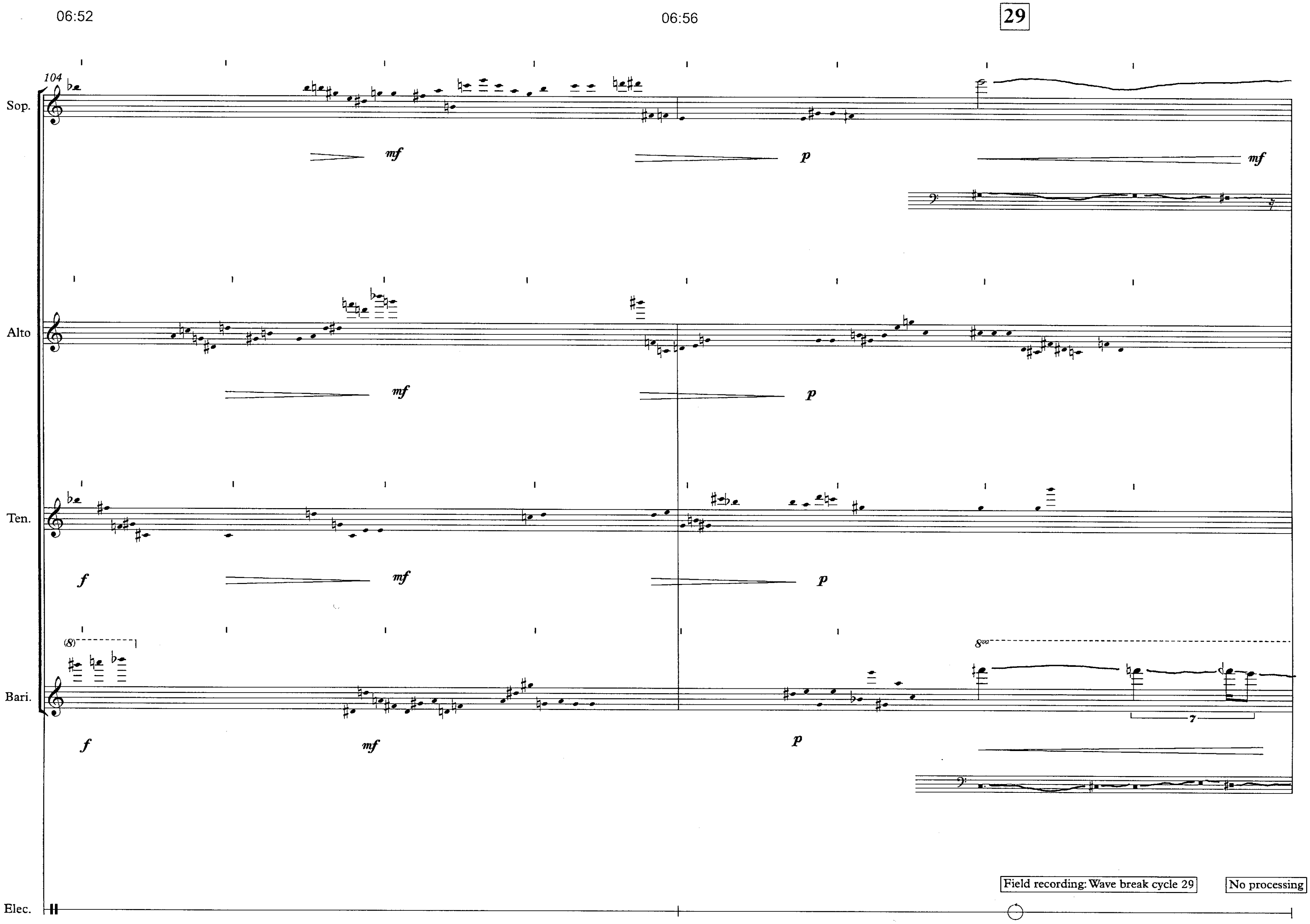

06:00
06:04
06:08
06:12
25
91
Sop.
Alto
Ten.
Bari.
Elec.
Field recording: Wave break cycle 25

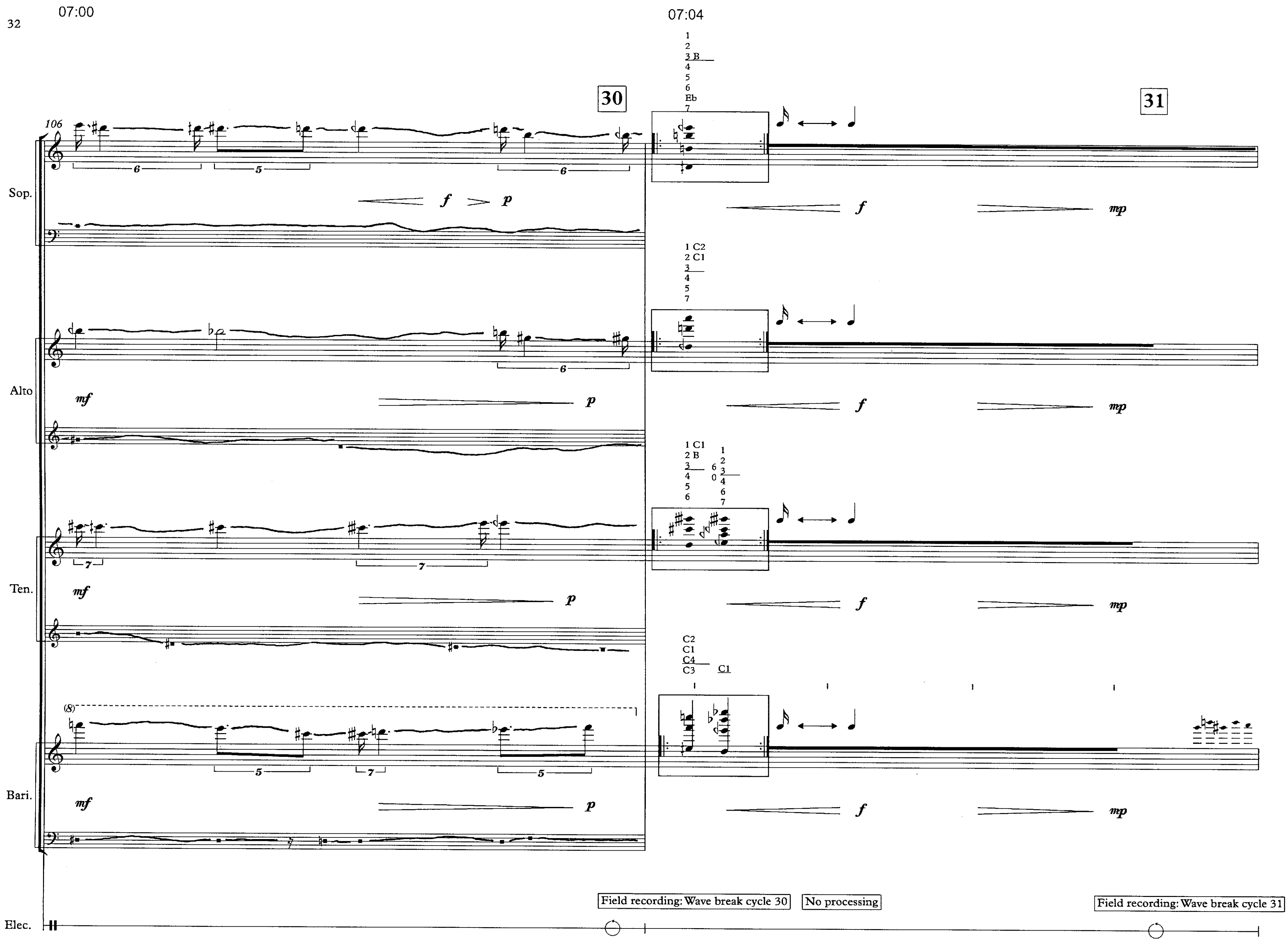

06:16
06:20
95
Sop.
Alto
Ten.
Bari.
Elec.
mf
f
No processing

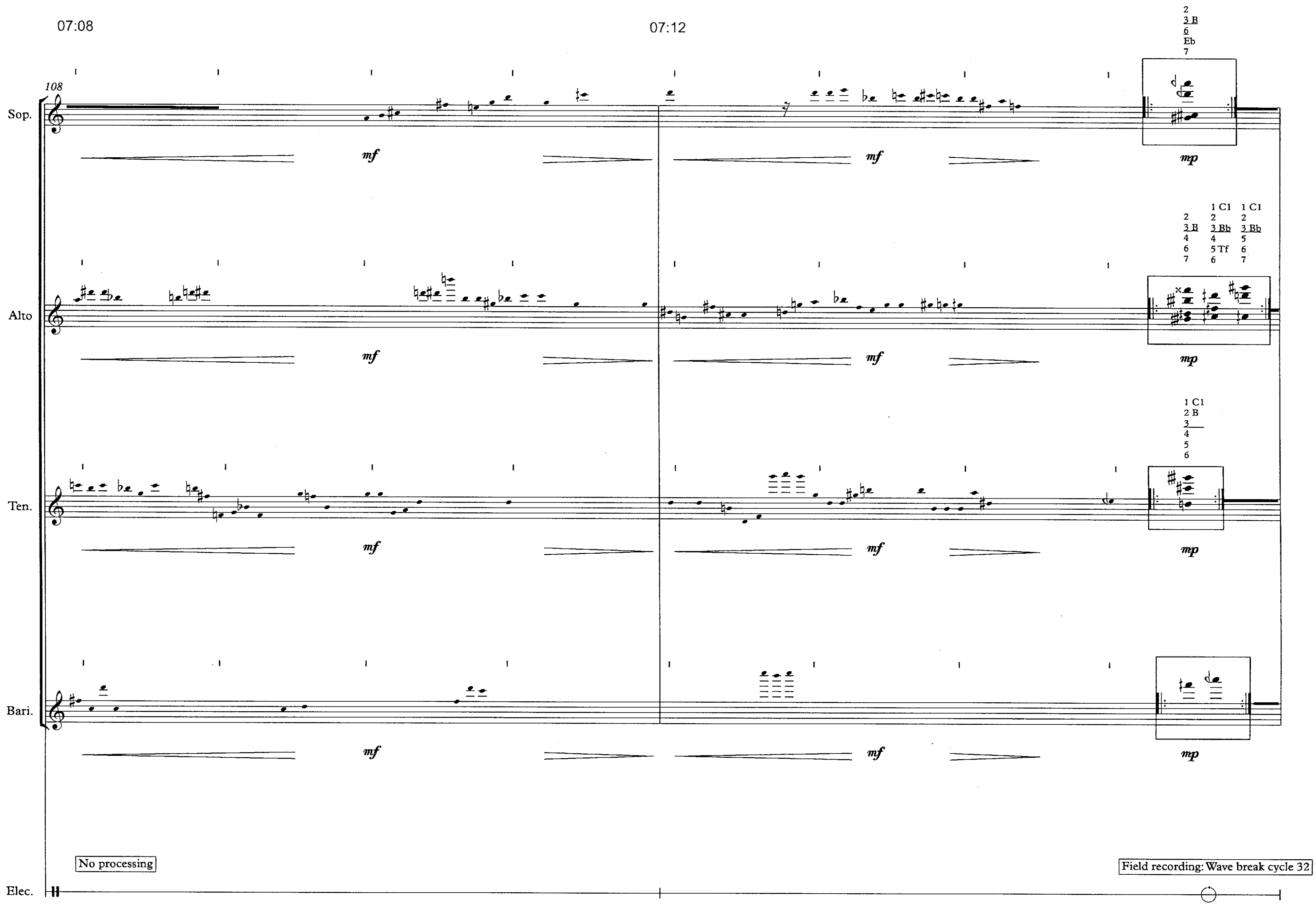

06:24
26
06:28
97
Sop.
Alto
Ten.
Bari.
Elec.
p
mp
8va
3
5
7
Field recording: Wave break cycle 26
Saxophone processing: pitch shifting / filtering

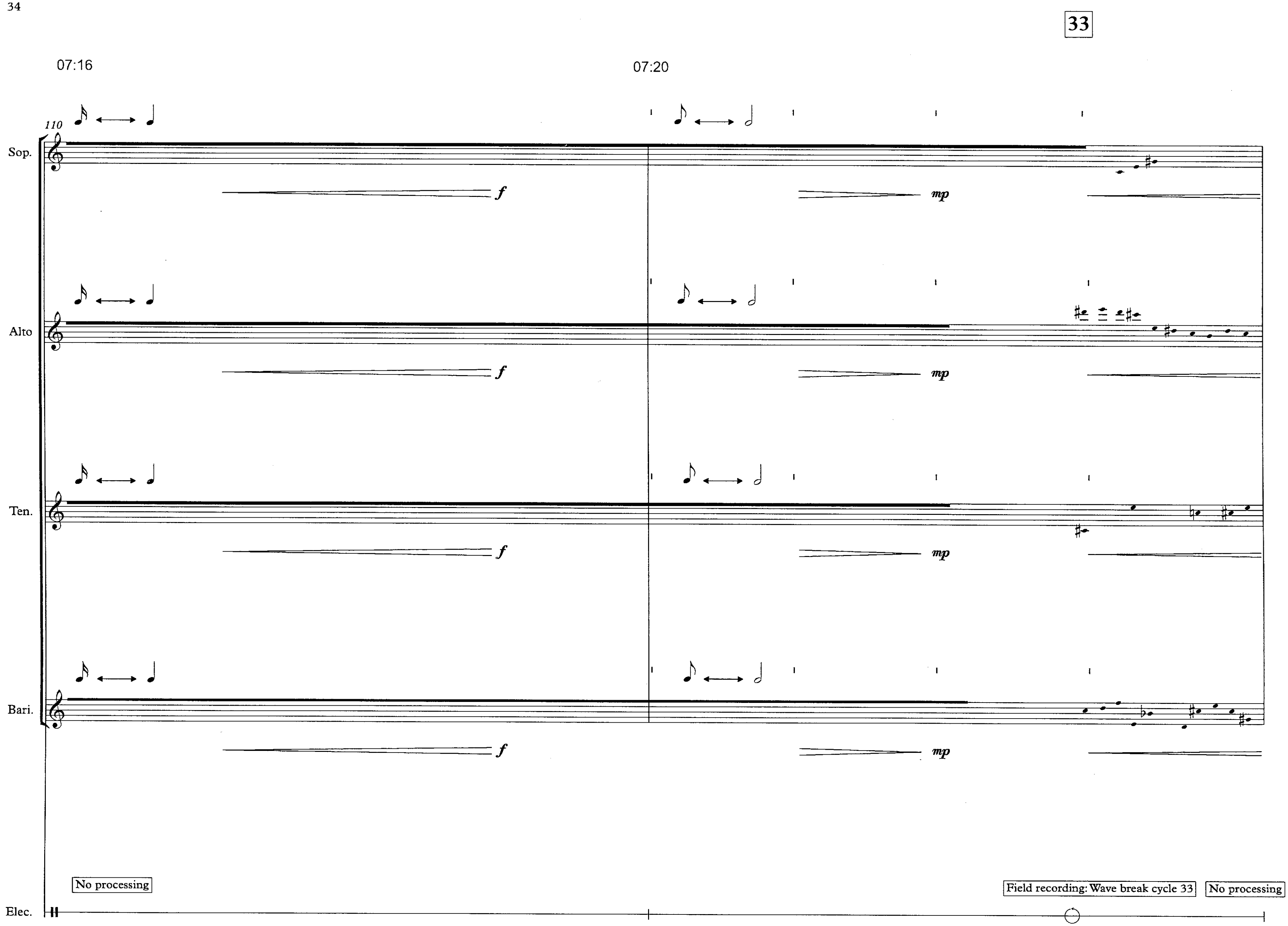

06:32
06:36
06:40
27
99
Sop.
p mp p
mf
Alto
p mp p
mf
Ten.
p mp p
mf
(8)
Bari.
mf
Field recording: Wave break cycle 27
Saxophone processing: pitch shifting / filtering
Elec.

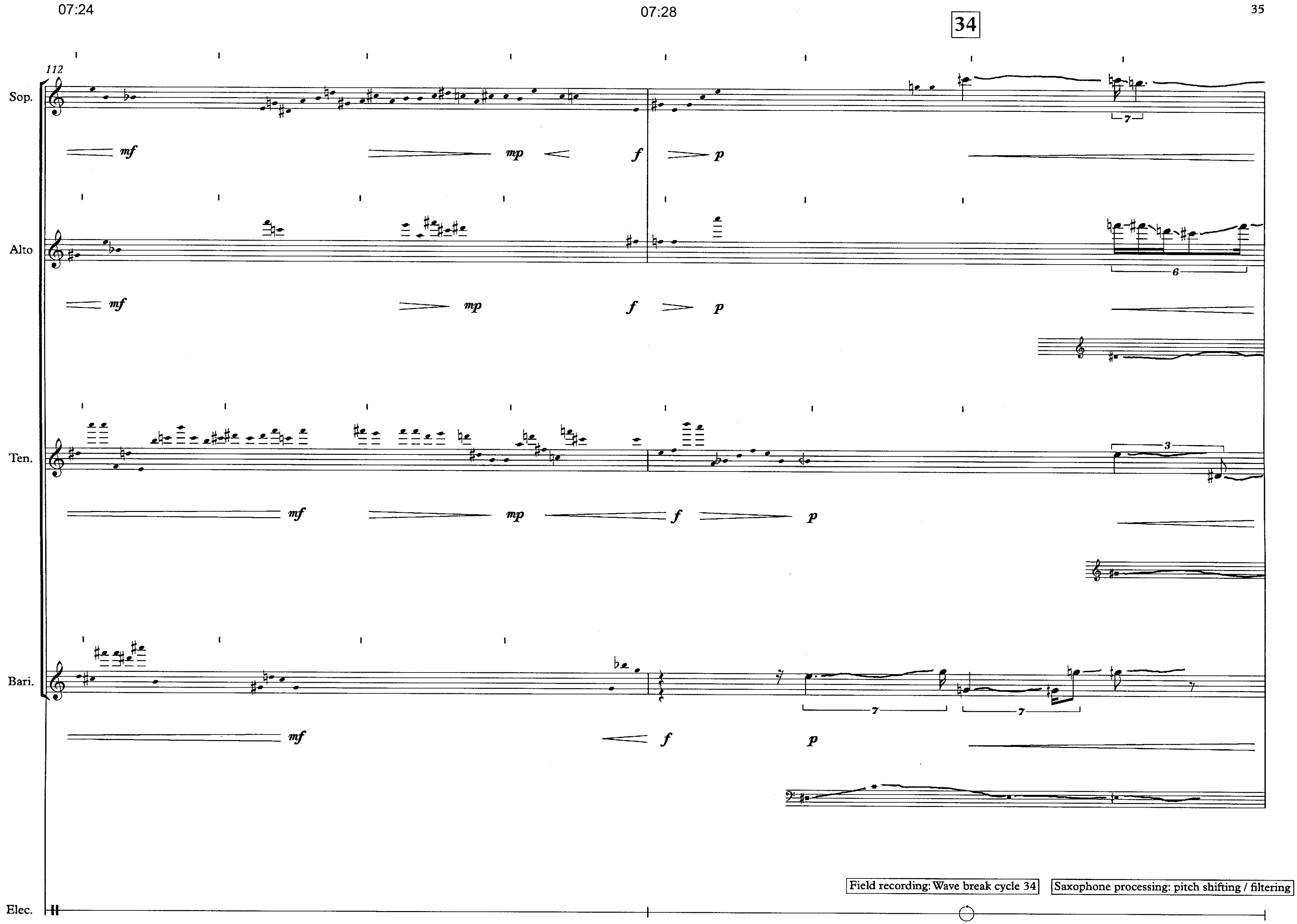

06:44
06:48
28
102
Sop.
Alto
Ten.
Bari.
(8)
Elec.
Field recording: Wave break cycle 28
No processing

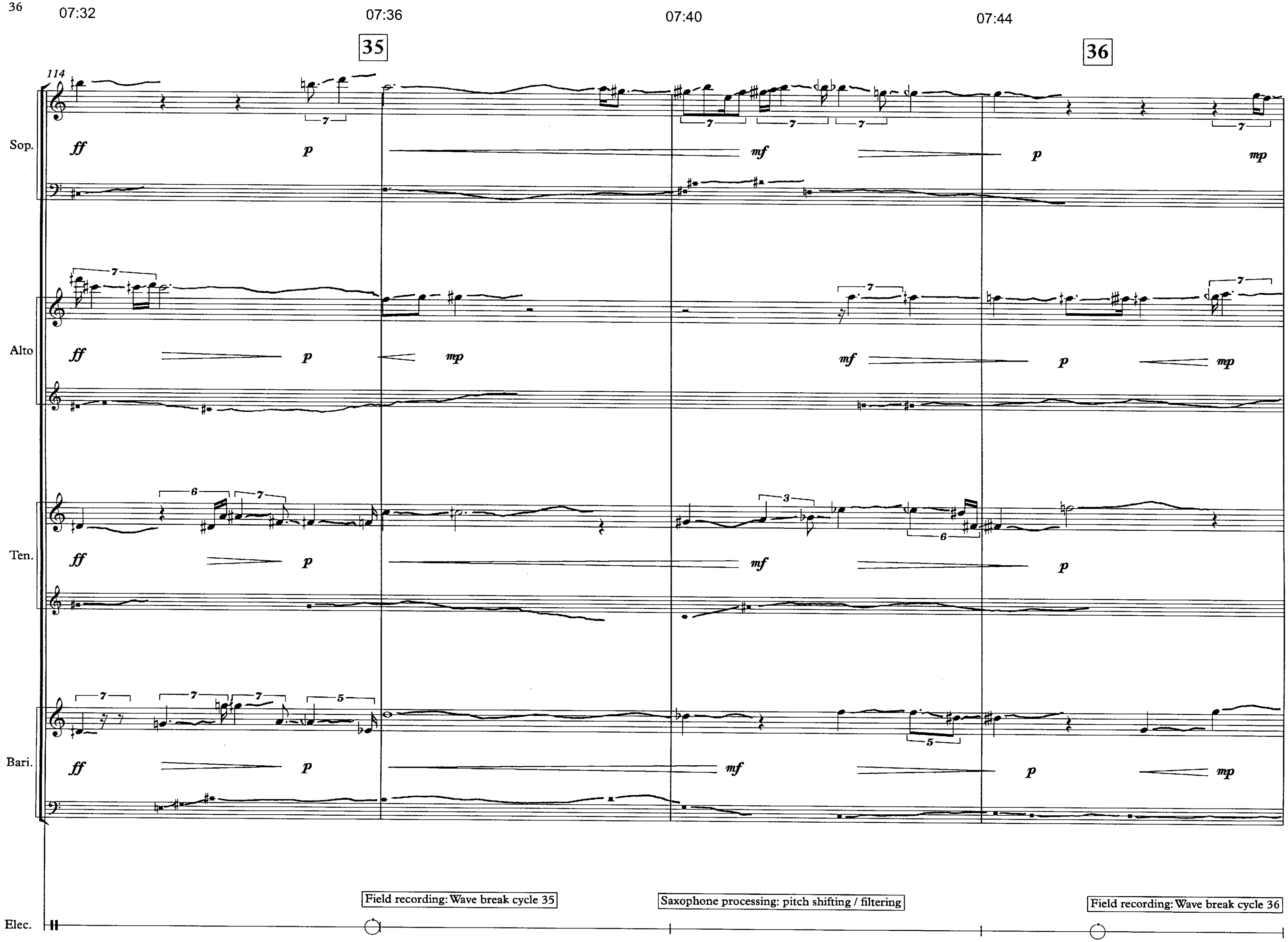

06:52
06:56
29
104
Sop.
Alto
Ten.
Bari.
Elec.
mf
p
f
(8)
8va
7
Field recording: Wave break cycle 29
No processing

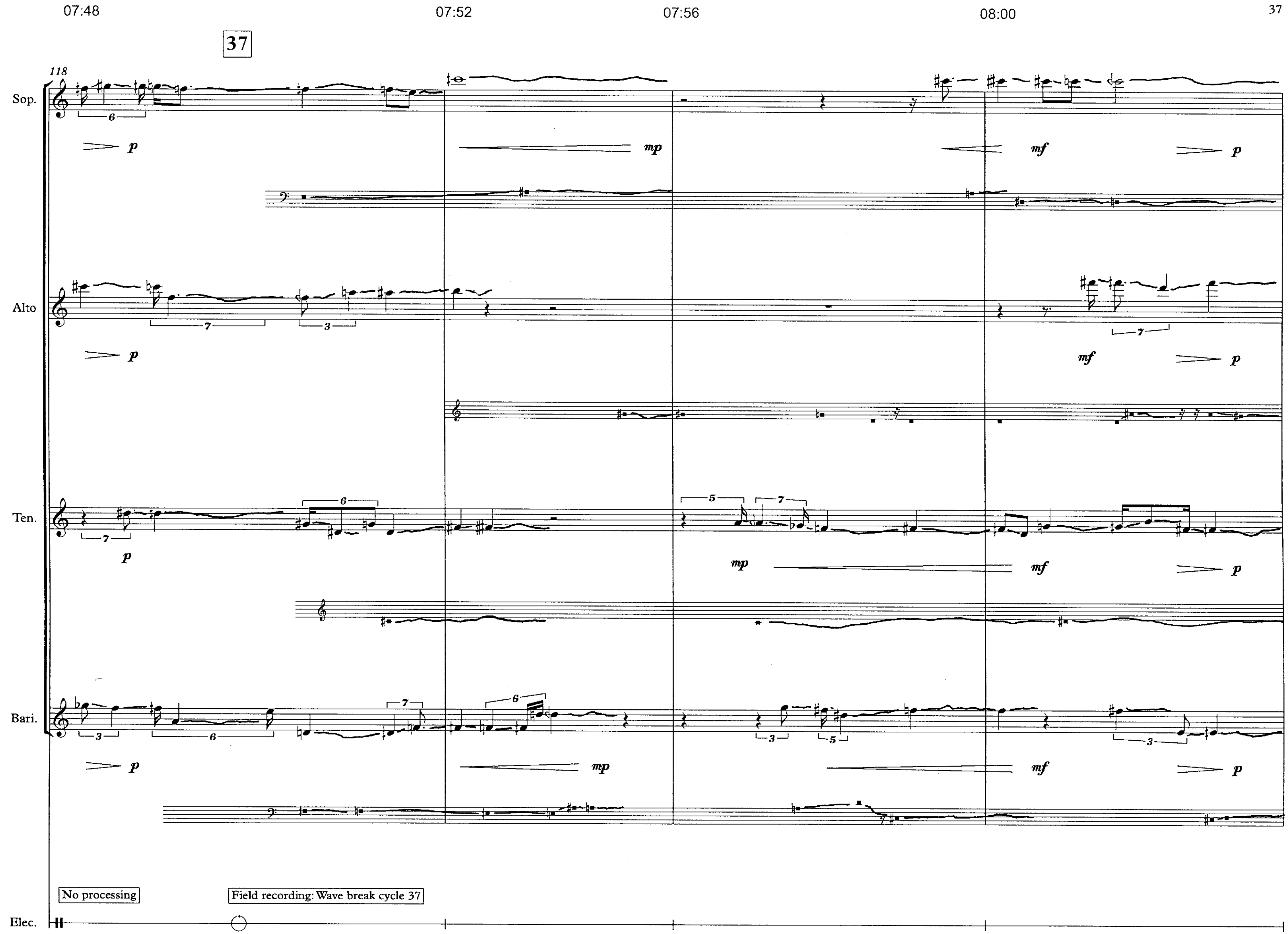

07:00
07:04
30
31
106
Sop.
Alto
Ten.
Bari.
Elec.
f
p
mf
mp
Field recording: Wave break cycle 30
No processing
Field recording: Wave break cycle 31

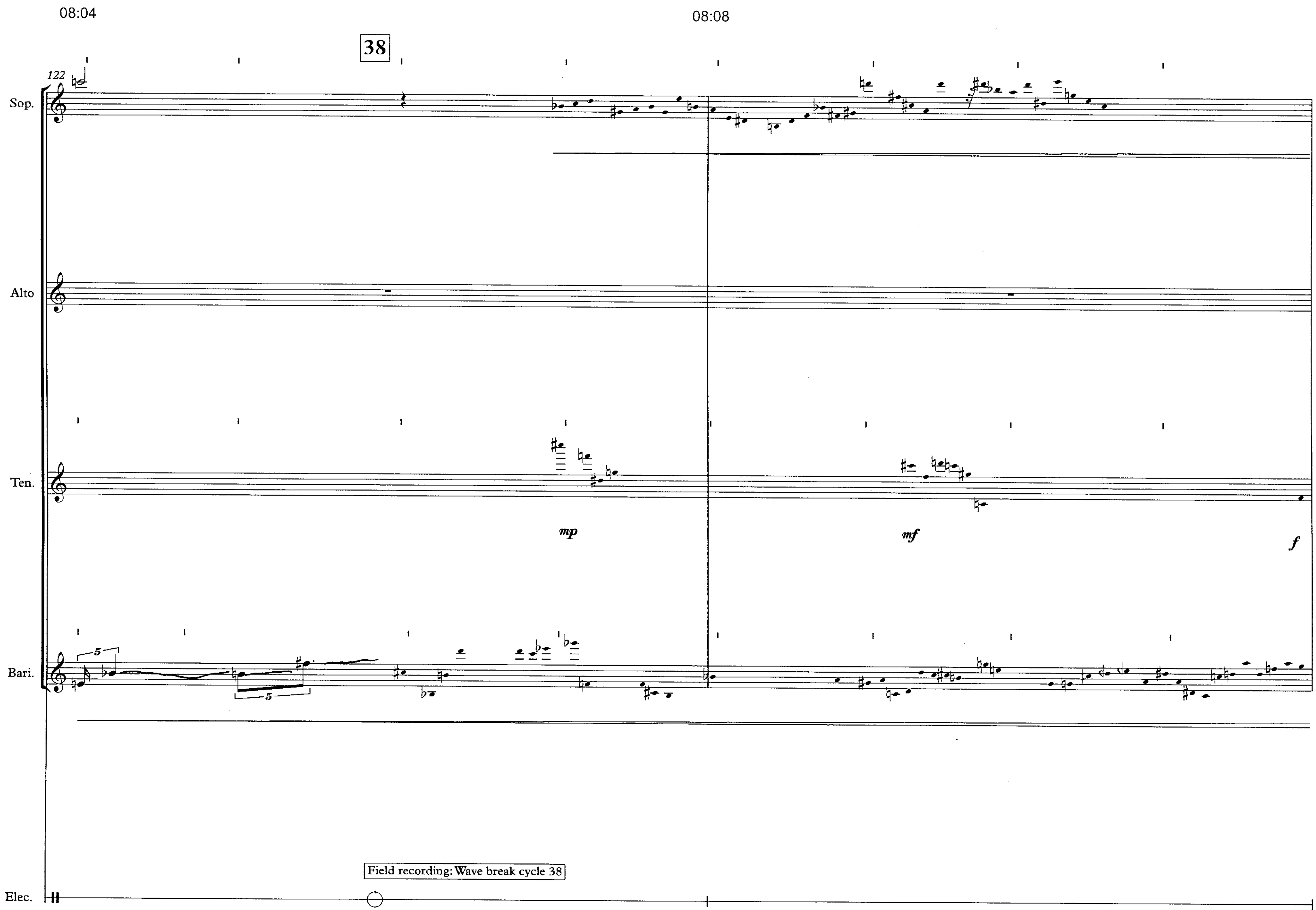

07:08
07:12
108
Sop.
Alto
Ten.
Bari.
Elec.
mf
mp
2 3 B 6 Eb 7
2 3 B 4 6 7
1 C1 2 3 Bb 4 5 Tf 6
1 C1 2 3 Bb 5 6 7
1 C1 2 B 3 4 5 6
No processing
Field recording: Wave break cycle 32

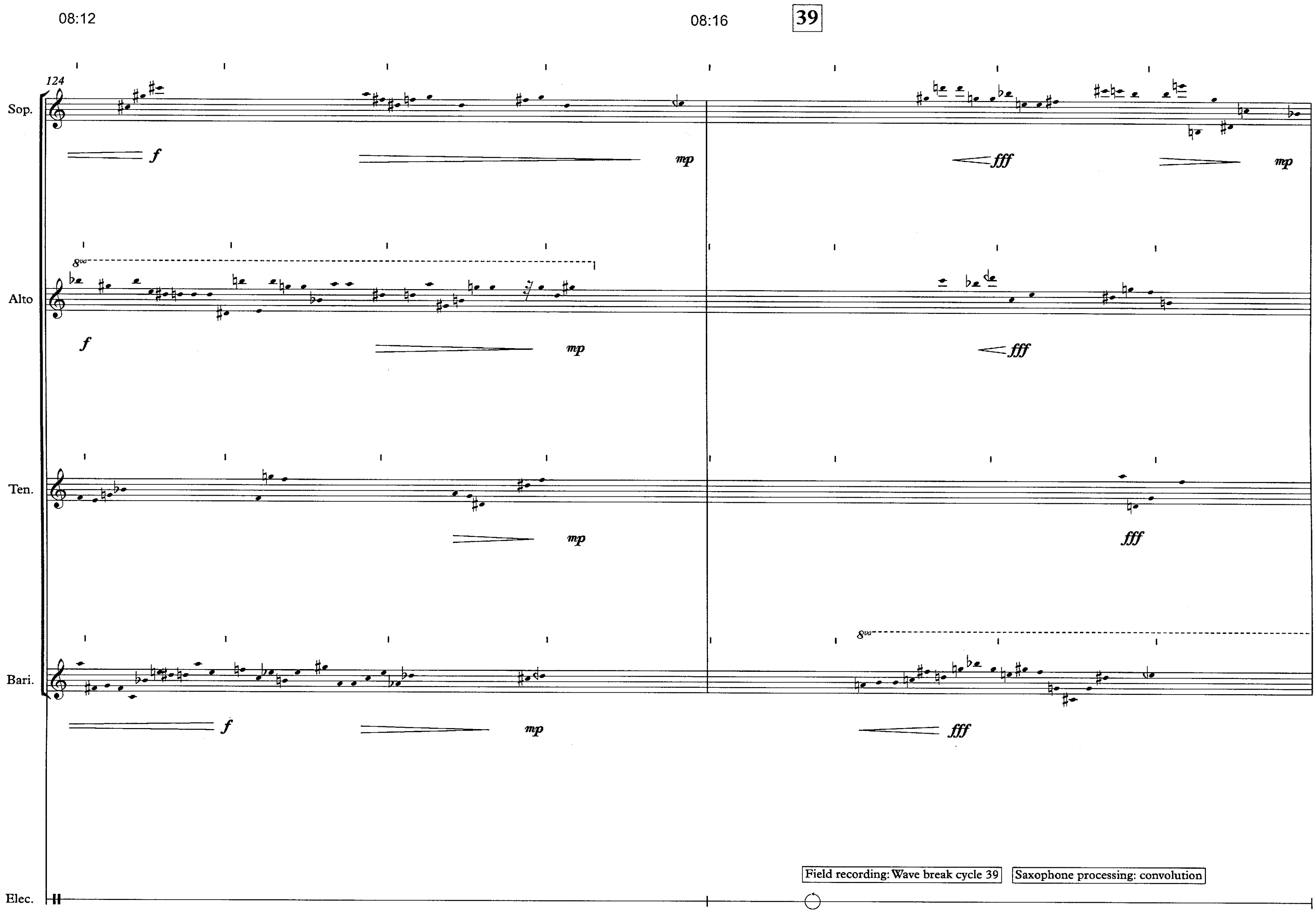

33
07:16
07:20
110
Sop.
Alto
Ten.
Bari.
Elec.
f
mp
No processing
Field recording: Wave break cycle 33
No processing

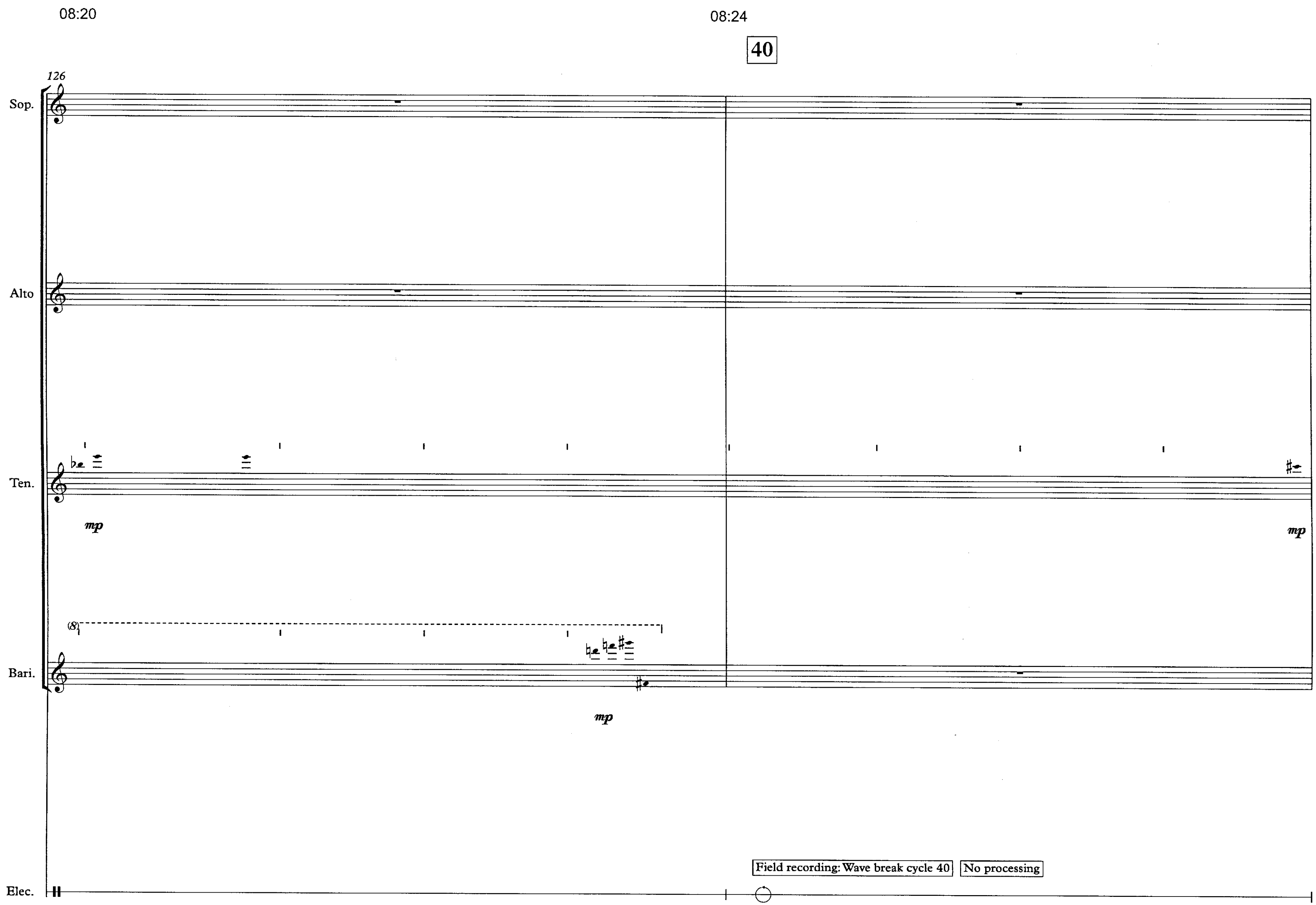

07:24
07:28
34
112
Sop.
mf
mp
f
p
7
Alto
mf
mp
f
p
6
Ten.
3
mf
mp
f
p
Bari.
7
7
mf
f
p
Field recording: Wave break cycle 34
Saxophone processing: pitch shifting / filtering
Elec.

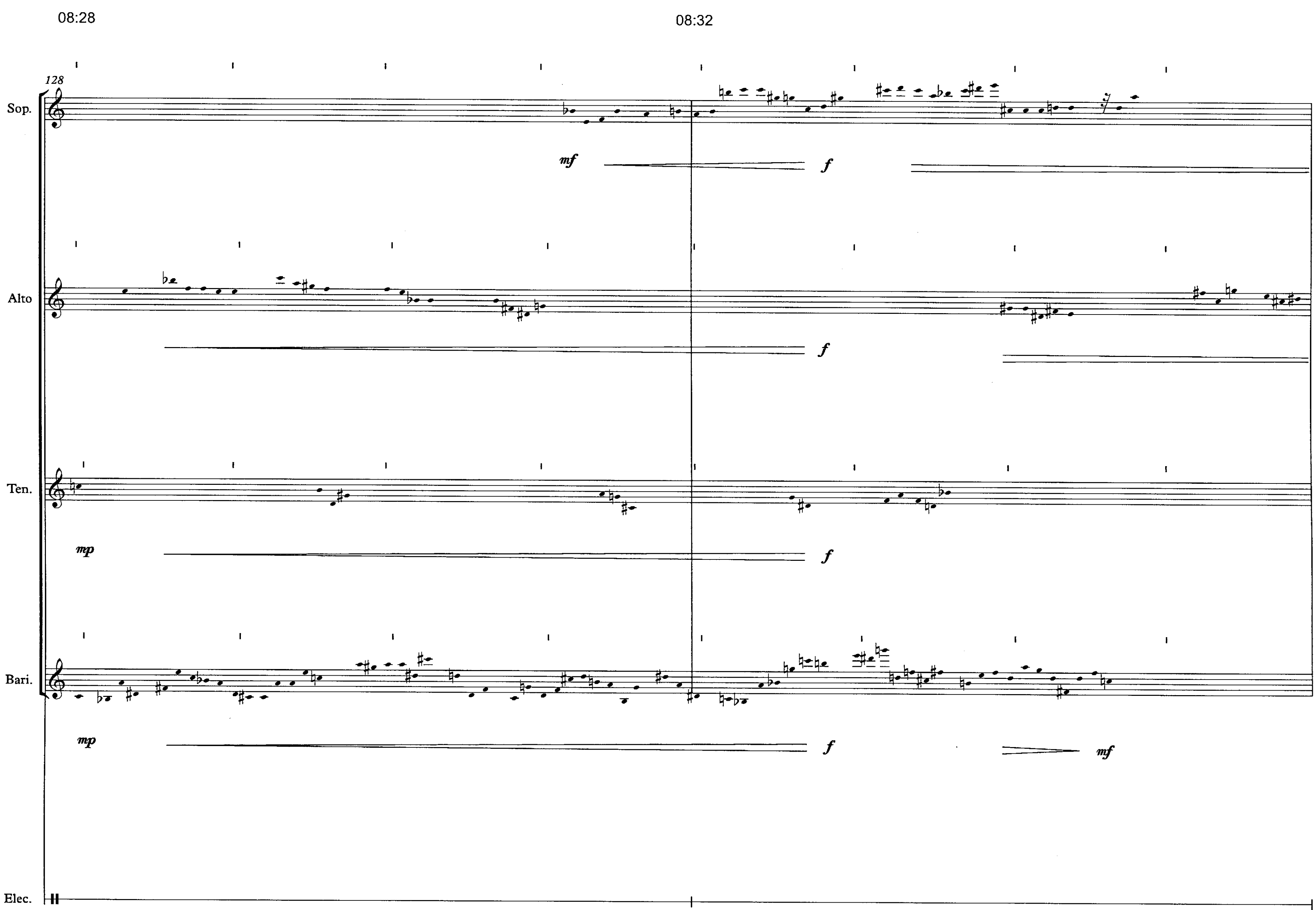

07:32
07:36
07:40
07:44
35
36
114
Sop.
Alto
Ten.
Bari.
Elec.
Field recording: Wave break cycle 35
Saxophone processing: pitch shifting / filtering
Field recording: Wave break cycle 36

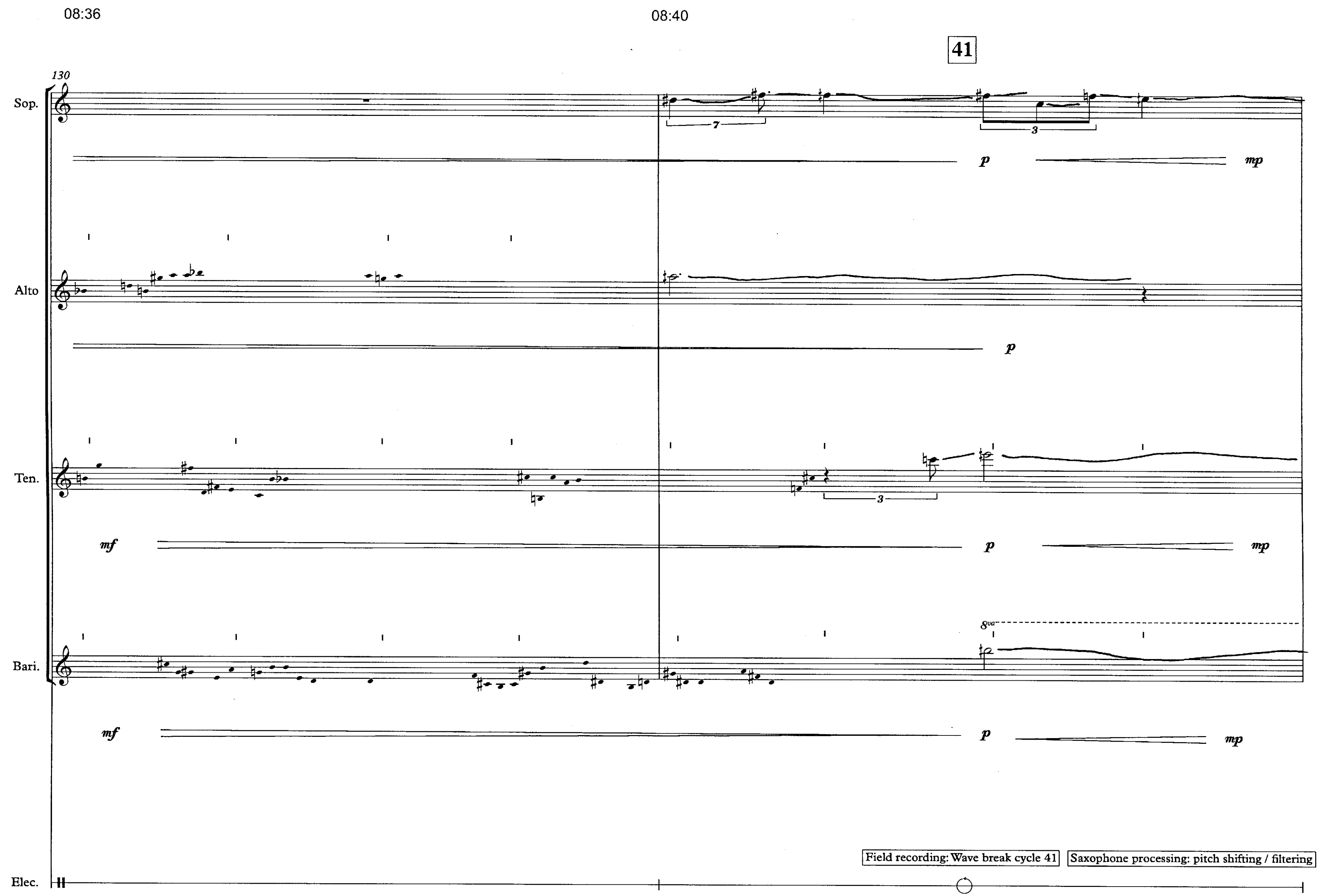

07:48
07:52
07:56
08:00
37
118
Sop.
Alto
Ten.
Bari.
Elec.
p
mp
mf
No processing
Field recording: Wave break cycle 37

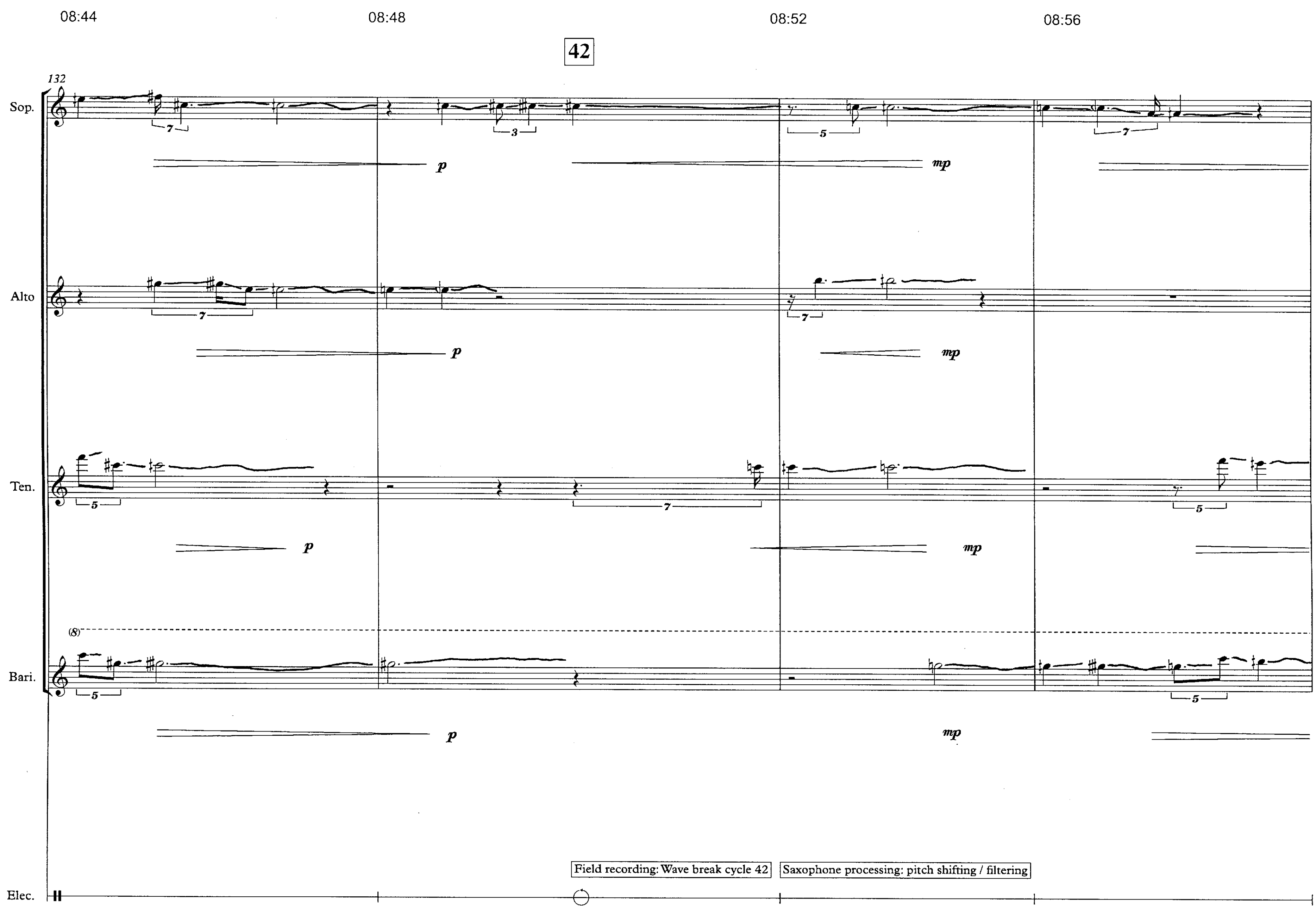

08:04
08:08
38
122
Sop.
Alto
Ten.
mp
mf
f
Bari.
5
5
Field recording: Wave break cycle 38
Elec.


08:12
08:16
39
124
Sop.
Alto
Ten.
Bari.
Elec.
f
mp
fff
8va
Field recording: Wave break cycle 39
Saxophone processing: convolution

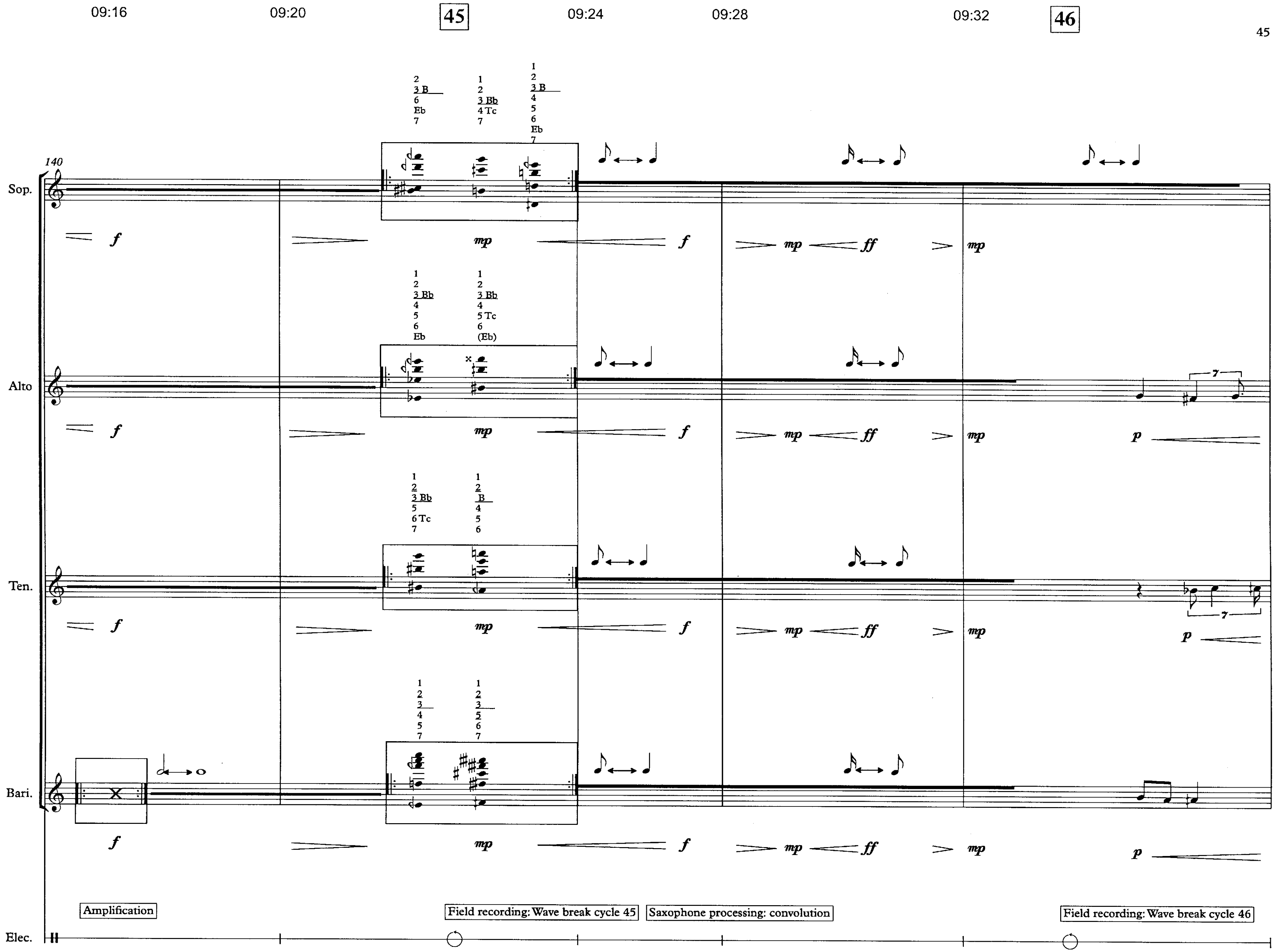

08:20
08:24
40
126
Sop.
Alto
Ten.
Bari.
mp
(8)
mp
mp
Field recording: Wave break cycle 40
No processing
Elec.

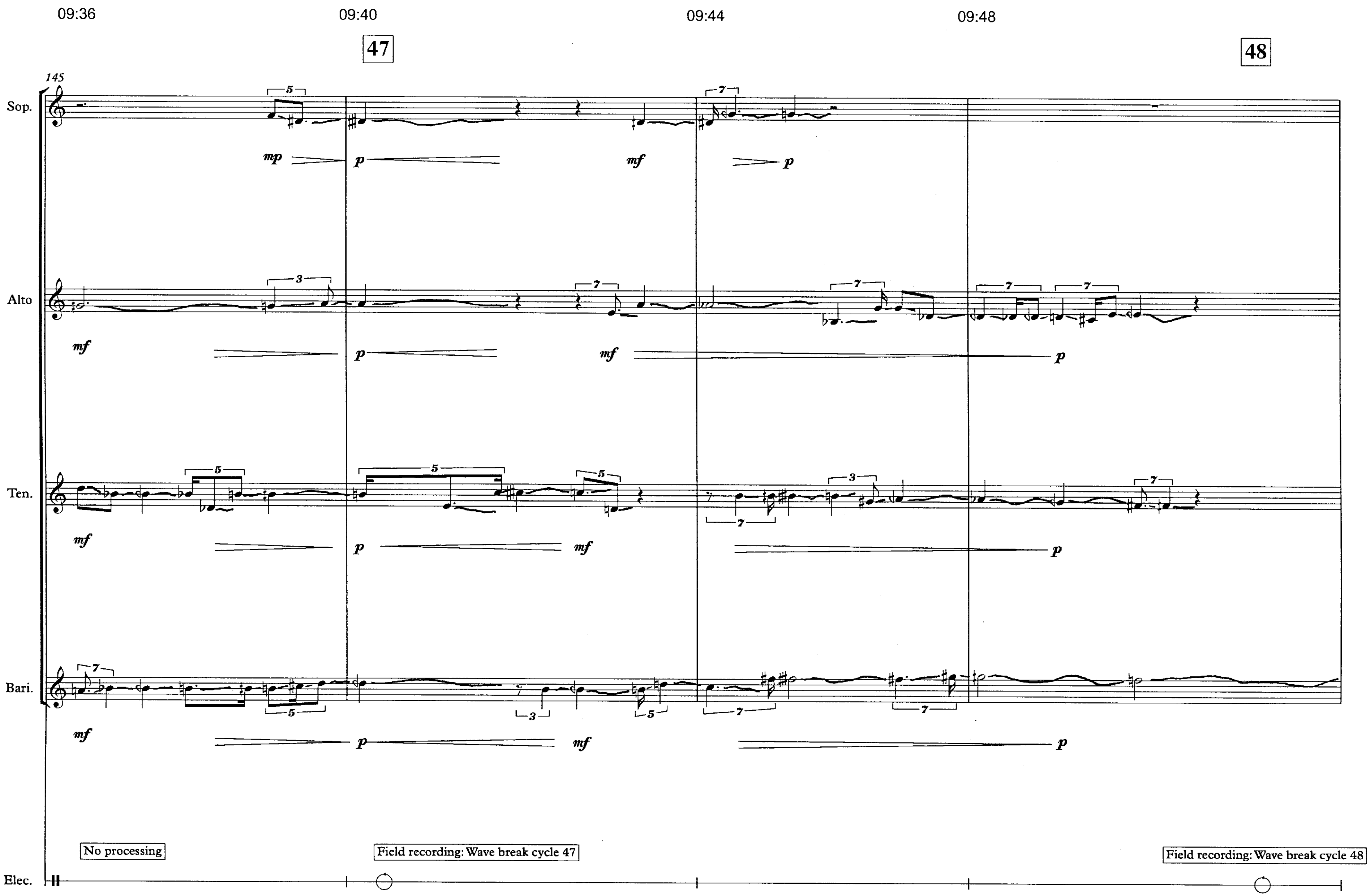

08:28
08:32
128
Sop.
Alto
Ten.
Bari.
Elec.
mf
f
f
mp
f
mp
f
mf


08:36
08:40
41
130
Sop.
Alto
Ten.
Bari.
Elec.
p
mp
mf
8va
Field recording: Wave break cycle 41
Saxophone processing: pitch shifting / filtering

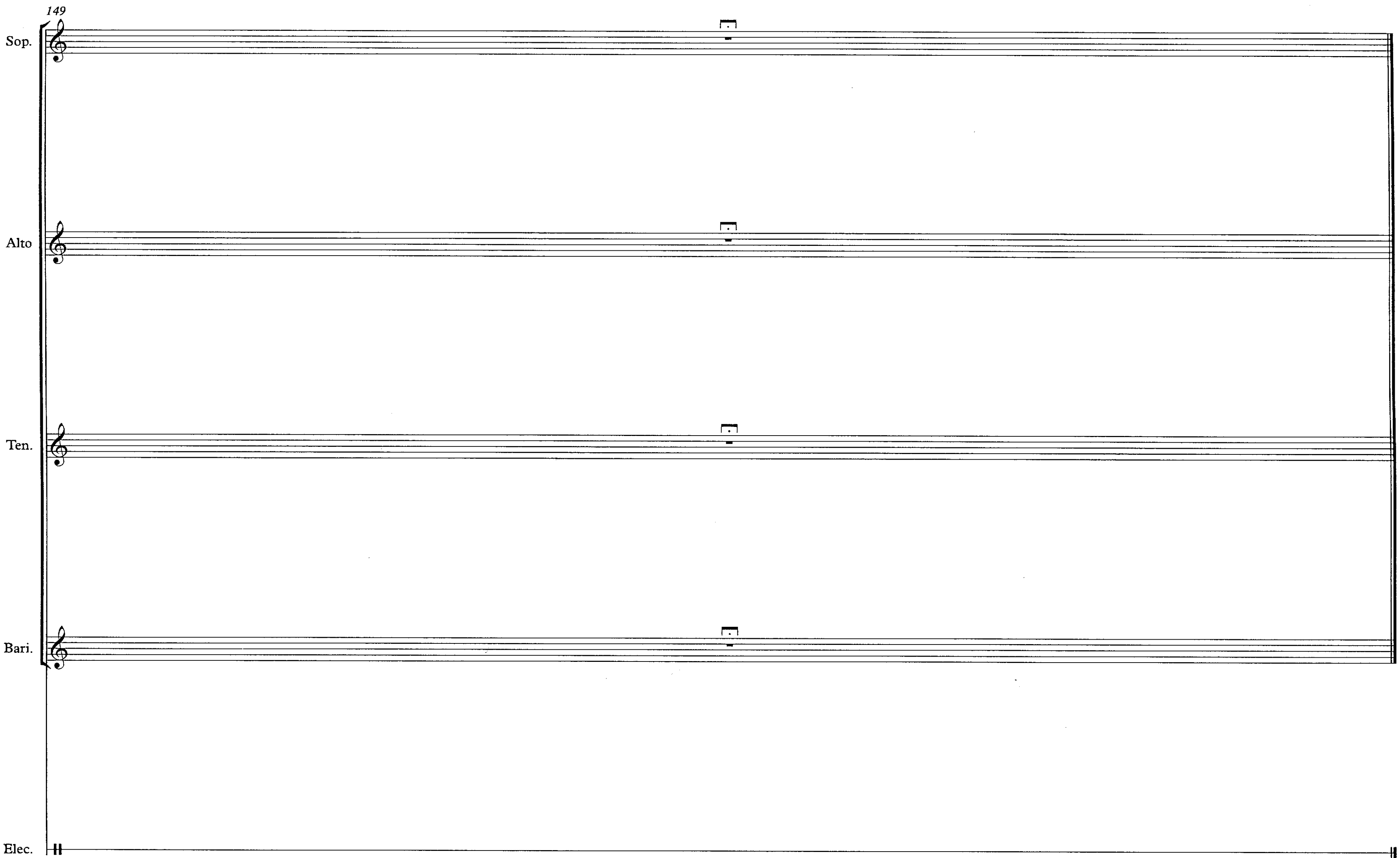

08:44
08:48
08:52
08:56
42
132
Sop.
Alto
Ten.
Bari.
Elec.
p
mp
Field recording: Wave break cycle 42
Saxophone processing: pitch shifting / filtering


09:00
43
09:04
09:08
09:12
44
136
Sop.
Alto
Ten.
Bari.
Elec.
Field recording: Wave break cycle 43
No processing
Field recording: Wave break cycle 44


09:16
09:20
45
09:24
09:28
09:32
46
140
Sop.
Alto
Ten.
Bari.
Elec.
Amplification
Field recording: Wave break cycle 45
Saxophone processing: convolution
Field recording: Wave break cycle 46


09:36
09:40
09:44
09:48
47
48
145
Sop.
Alto
Ten.
Bari.
Elec.
No processing
Field recording: Wave break cycle 47
Field recording: Wave break cycle 48

09:52



Night music

summer dusk and night sounds
of insects and other creatures in Virginia forests

for Splinter Reeds

Christopher Luna-Mega (2018)

Proportional Notation / Durations

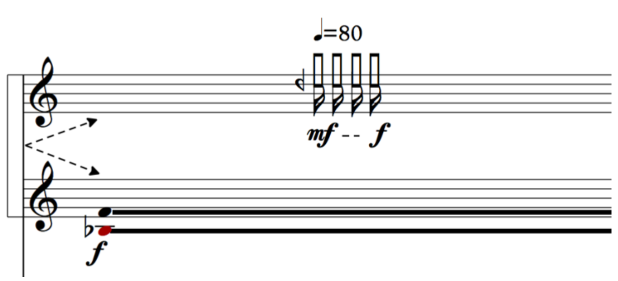
Each system is 10 seconds long, divided into two 5 second segments. Sounds must be played according to their position relative to the timeline and the length of the beam that follows a note-head. Note-heads without a duration beam must be played as short as possible. Stems do not function as rhythmic values, but as connections of groups of notes that immediately follow one another.

Simultaneous Material / Staff Choices

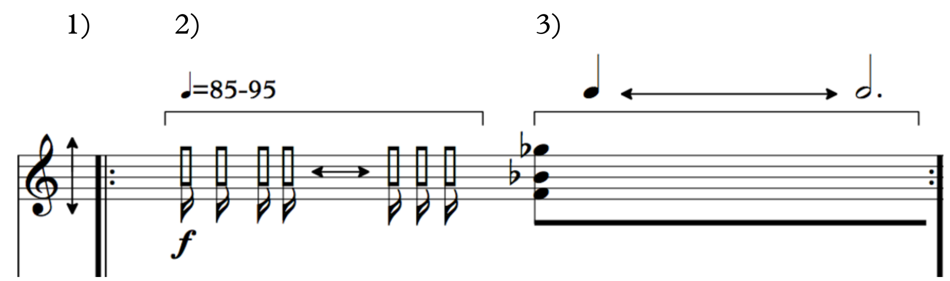
In most sections of the piece, each performer reads from two simultaneous staves. Some sections present juxtaposed material from staff to staff, in which the performer alternates between staves without leaving material out of the performance. When a section presents simultaneous material in both staves, the performer chooses between both materials, trying to cover as much of the material written in both staves.

<u>Dashed arrows facing both staves</u>

Presented when material of one staff is sustained and another staff is brief. The sustained material on one staff must be interrupted and yield to the brief material when the latter appears on the other staff. As soon as the brief material on one staff is played, the sustained material on the other staff must be resumed. The alternation points must be used for breathing when needed. This procedure takes place in the opening sections of movements 1 and 4.



Indeterminate material cells



1) <u>The upward/downward facing arrow</u> applies to both the noised rectangular note-heads (see explanation in the following page) and the round note-heads presented as a chord. For noised rectangular note-heads, constantly vary the register; for the defined pitches, choose one of them on every repetition.

2) <u>Imitative noise rhythmic choices.</u> Choose between the two rhythms at the given approximate tempo. After playing the chosen motive, immediately or after a brief pause for breathing move on to the pitched material.

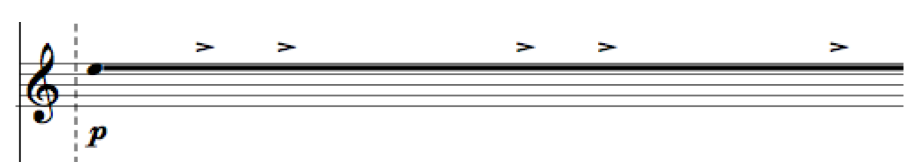
3) <u>Pitched material duration.</u> Play a single chosen pitch with any of the given durations. After playing the chosen pitch, repeat the indeterminate material cell until the next time cue.

Dynamics

1) ***p* --------- *mf***

Constantly vary within the given dynamic range.

2)

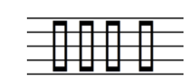


Accent without re-articulating the note.

Accidentals

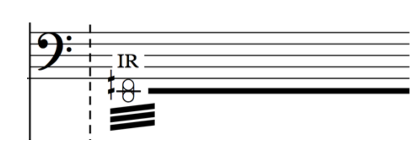
1/8 tone up	1/8 tone down
1/4 tone up	1/4 tone down
3/8 tone up	3/8 tone down
1/2 tone up	1/2 tone down
5/8 tone up	5/8 tone down
3/4 tone up	3/4 tone down

Imitative noise



Based on the particular vocabularies of the instrumentalist, the rectangular note-head motives must imitate a predominantly noisy sound that is part of the summer forest dusk recording from which all the materials in this piece were derived. The sound is prominent in audio files 3, 4 and 5 of the supplementary material. These sounds are not beamed, therefore they are as brief as possible. The metered equivalent of the spatial notation is three or four ♪ at ~90 bpm, consistent throughout the piece.

Tremolos



Include all the possible pitches within the given range in tremolo with irregular (IR) durations and patterns.

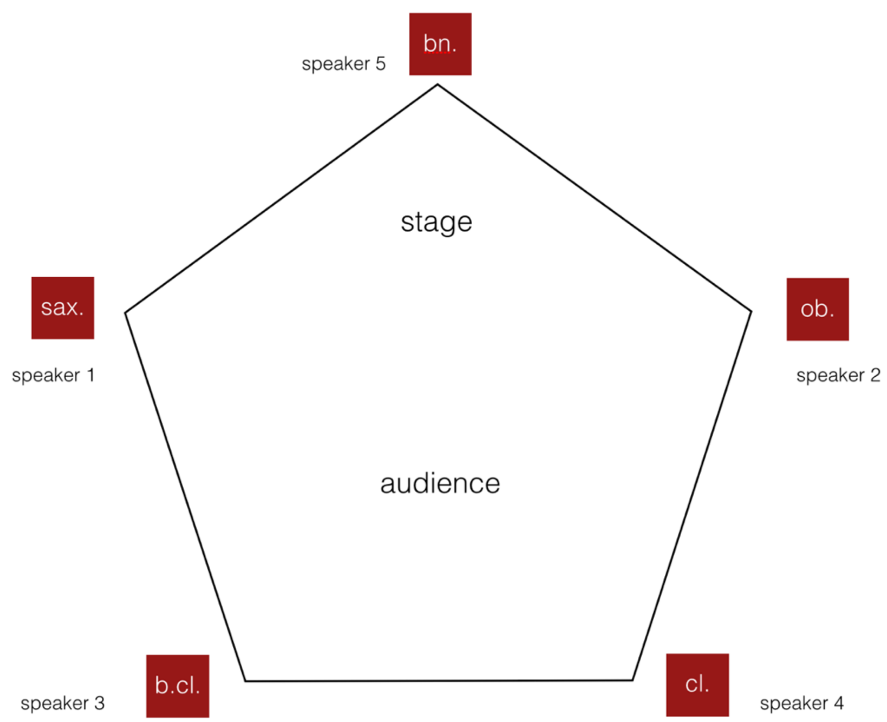
Two-pitch tremolo with irregular (IR) durations and patterns.

Spatial distribution, electronics and amplification

The piece may be performed with or without electronics/amplification, as well as with or without a spatial distribution of the performers.

Spatial distribution

The diagram below considers a standard concert hall. The spatial distribution of performers/speakers may be modified according to the venue.



Electronics and amplification

For the electronic version of the piece, all performers must be amplified. The electronics consist of field recordings and synthesized sounds. An audio file with the fixed media is available via e-mail: ch.luna.mega@gmail.com

Various versions of the piece are available (stereo, 5.1, 5.0, 4.1, 4.0).

Time cues and synchronization

The temporal organization of the piece is based on seconds/minutes, indicated throughout the score/parts substituting bar-numbers. A stopwatch is needed for each performer.

There are three movements in the piece, each requiring resetting the stopwatch as well as a synchronized onset conducted by a designated performer.

Supplementary material

All the musical materials performed by the reed quintet and used in the electronics are derived from direct transcriptions and arrangements of recordings of the summer dusk and night sounds of insects and other creatures in Virginia forests. Every movement in the piece is a fragment taken from the 40 minute original recordings. The striking increase in density and loudness as dusk becomes night is the guiding formal principle of the piece.

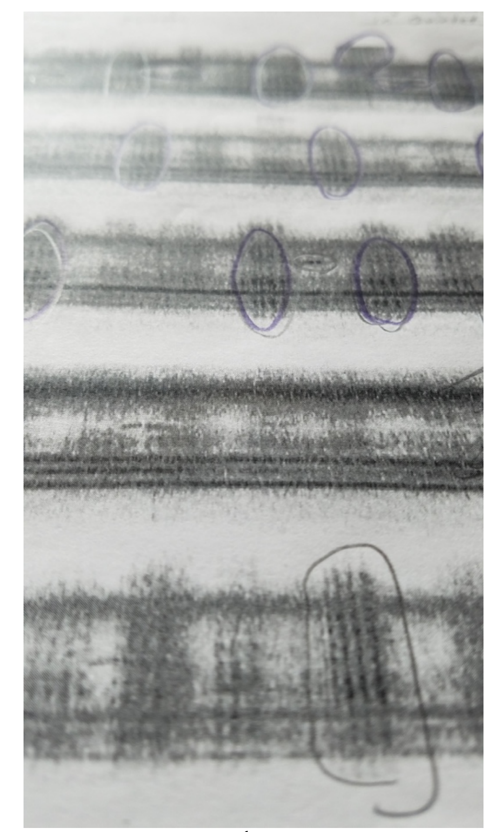
The recording uses five microphones in a pentagonal formation, at a distance of ~30 meters between each mic. Each of the five microphone analyses and transcriptions was assigned to an instrument (mic 1 to ob.; mic 2 to cl., etc.), rotating the pairings in each movement.

Included with the score and parts are the following fragments of the first 20" of the original field recordings:

1. nightmusic-fieldrec.-mvmt1 (sax / ob., cl.)
2. nightmusic-fieldrec.-mvmt2 (ob. / bn.)
3. nightmusic-fieldrec.-mvmt3 (sax. / b. cl.)
4. nightmusic-fieldrec.-mvmt4 (ob.)
5. nightmusic-fieldrec.-mvmt5 (*tutti*)

Please take a few moments prior to rehearsal to listen to these recordings while following along with the score (the beginning of each movement). The parentheses next to the audio files show the instruments that have been assigned to the featured sounds of the recording (first, the instrument that plays the sounds in the foreground; second, the instrument/s that play the sounds in the background). Besides the "imitative noise" notation explained in the previous page, an ideal performance of this piece includes the performers' input regarding instrumental techniques (besides those provided in the score) that best embody the sounds featured in the recordings.

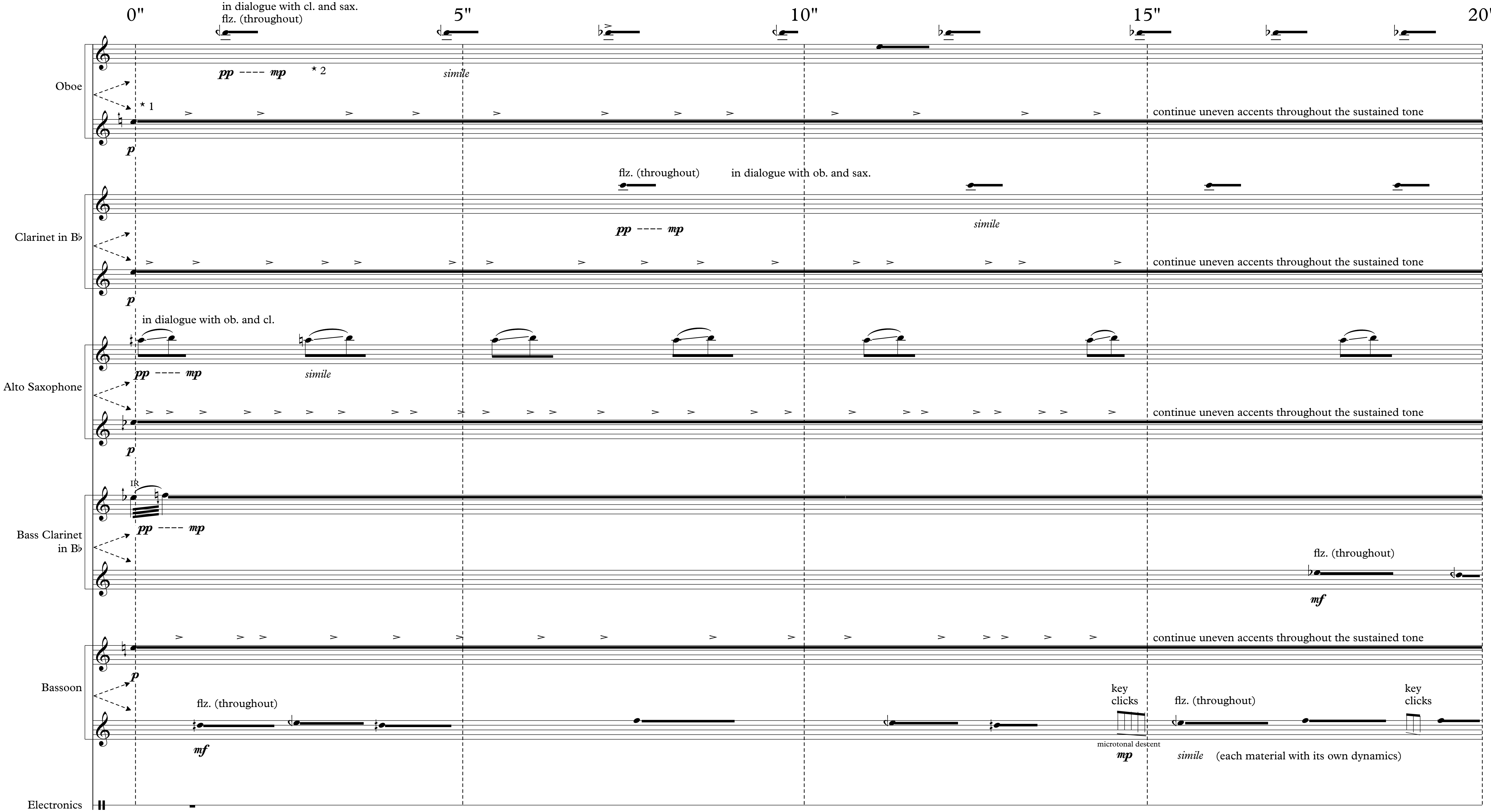
Microphone 4, Walnut Creek Park, Albemarle, VA, August 16, 7:15 pm.



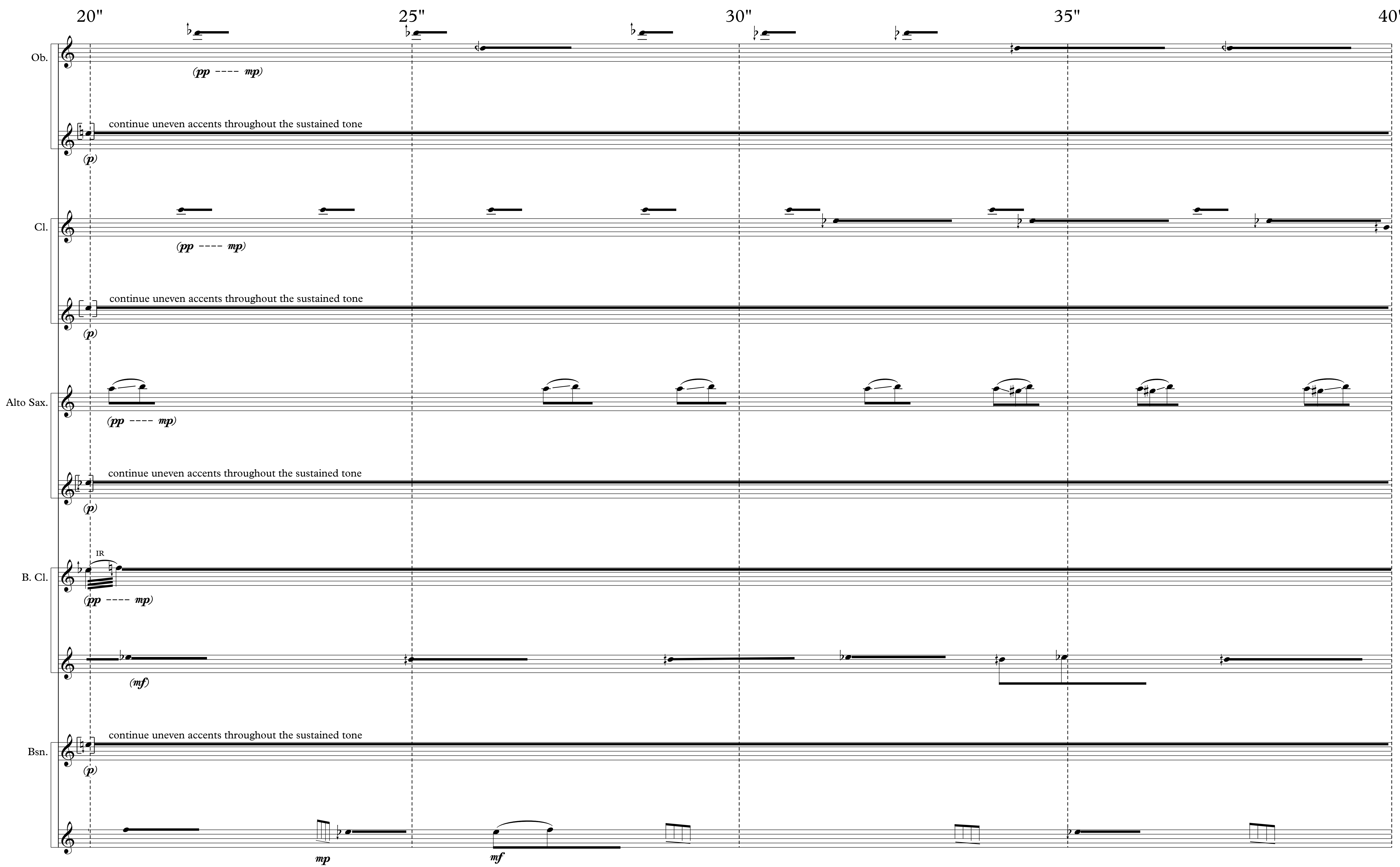
Sonogram of the 4th movement.

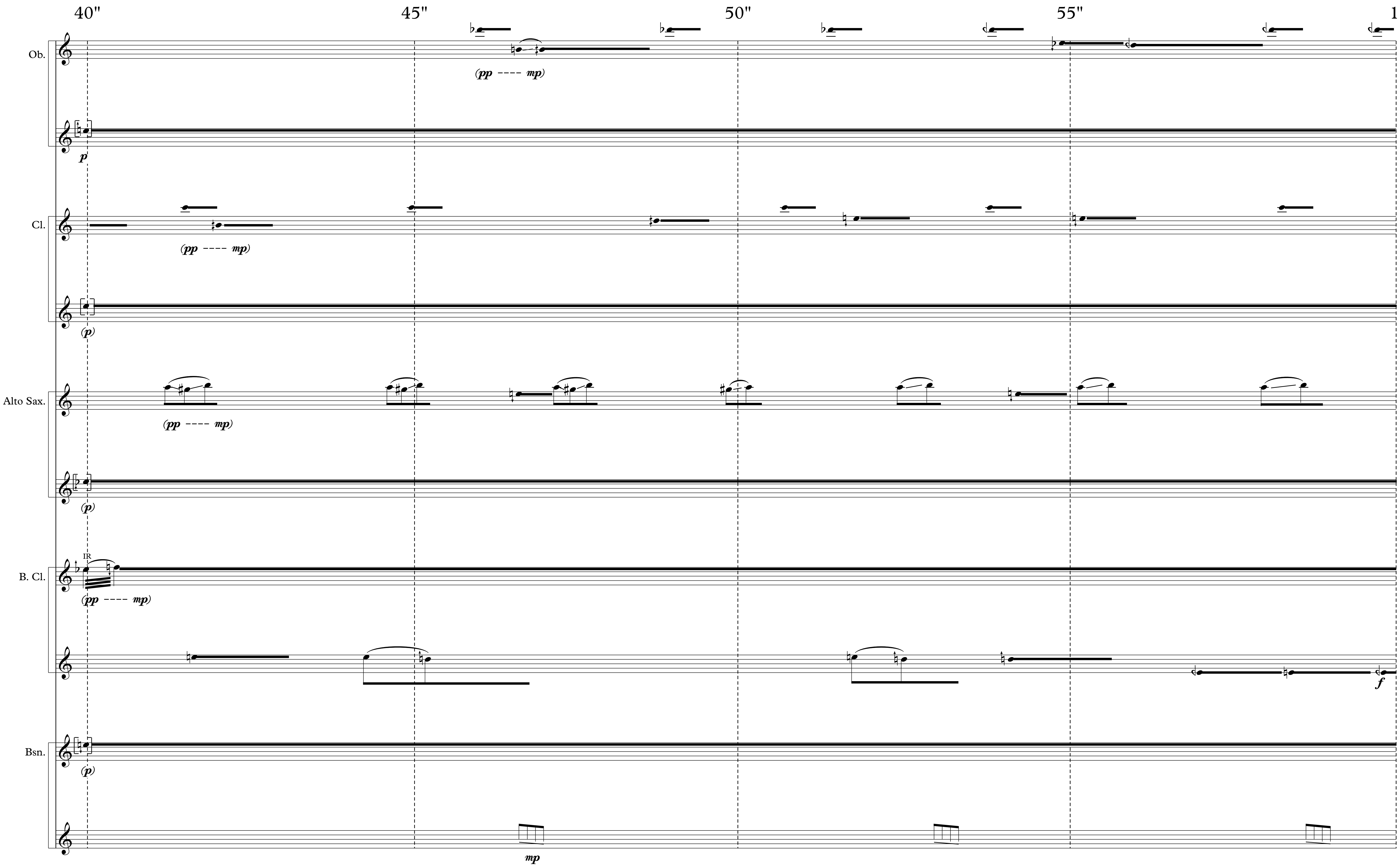
Night music
1.

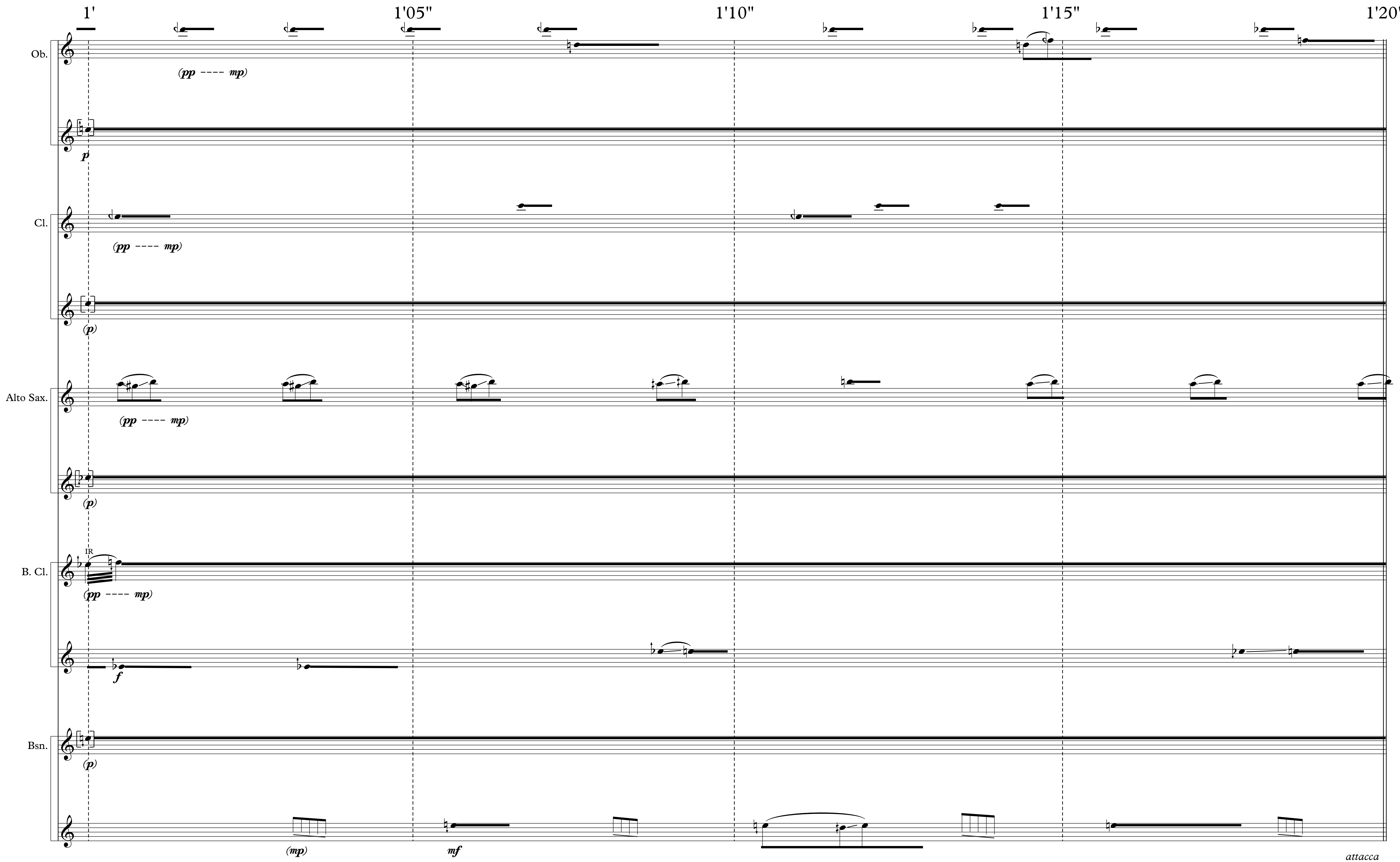
Christopher Luna-Mega (2018)



* 1) The sustained pitch material must sound at all times as a group (not individually) during this section.
* 2) Choose any dynamic within the given range.

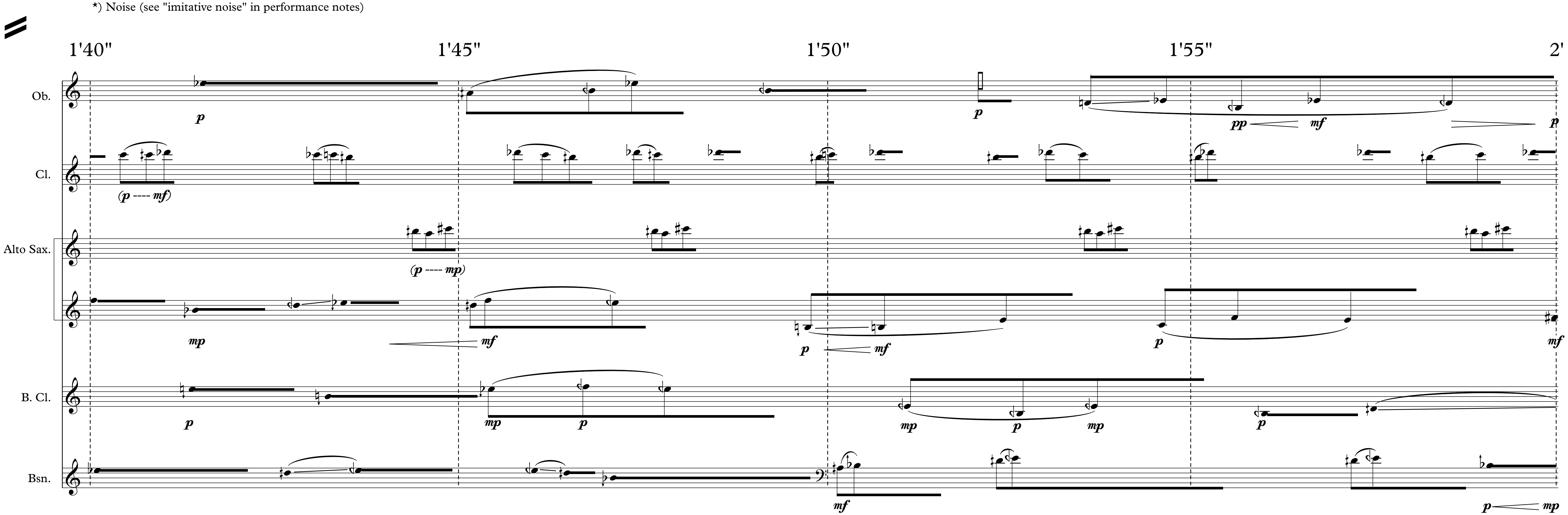

20"
25"
30"
35"
40"
Ob.
(pp ---- mp)
continue uneven accents throughout the sustained tone
(p)
Cl.
(pp ---- mp)
continue uneven accents throughout the sustained tone
(p)
Alto Sax.
(pp ---- mp)
continue uneven accents throughout the sustained tone
(p)
B. Cl.
IR
(pp ---- mp)
(mf)
Bsn.
continue uneven accents throughout the sustained tone
(p)
mp
mf

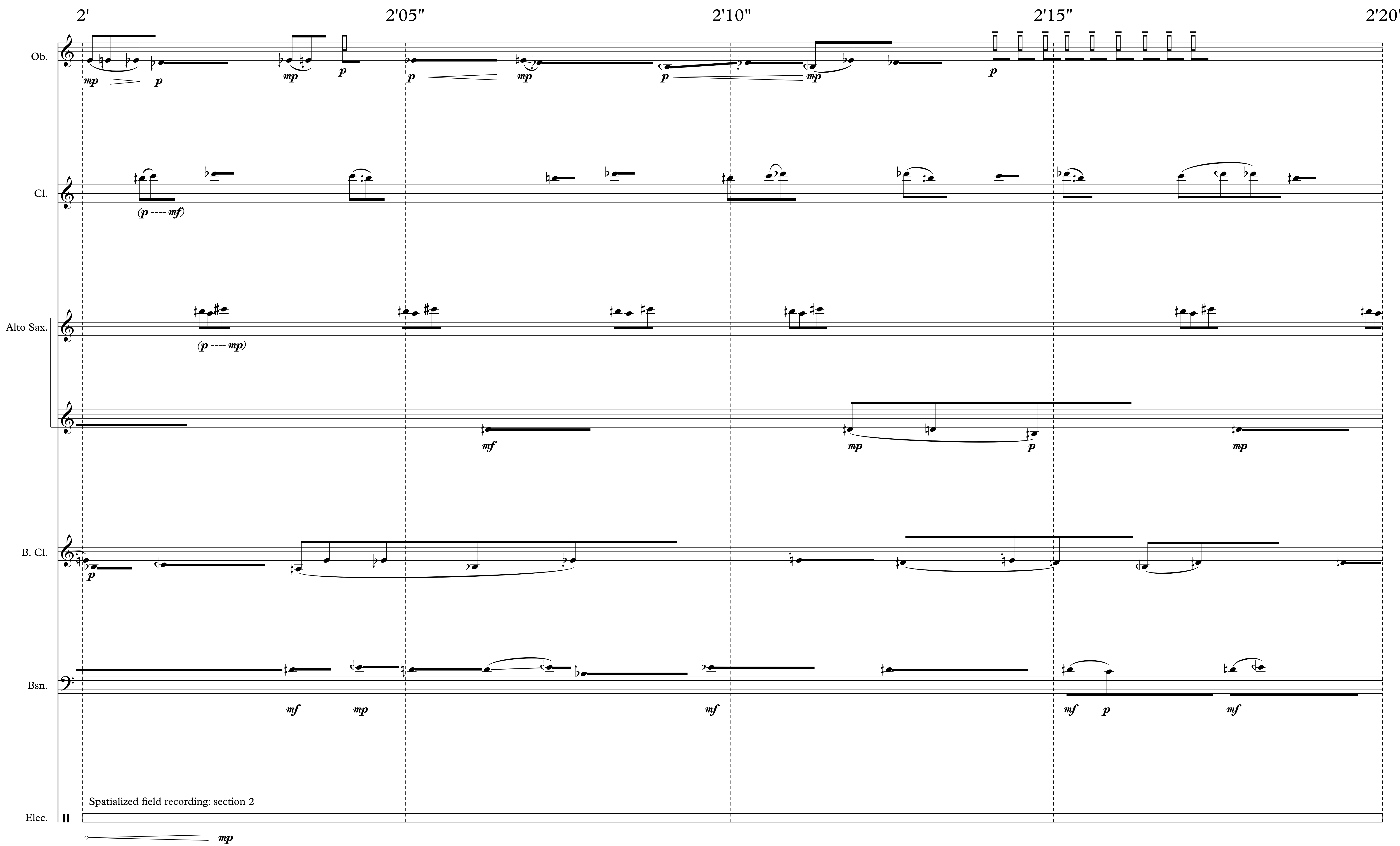

40"
45"
50"
55"
1'
Ob.
(pp ---- mp)
p
Cl.
(pp ---- mp)
(p)
Alto Sax.
(pp ---- mp)
(p)
B. Cl.
IR
(pp ---- mp)
f
Bsn.
(p)
mp

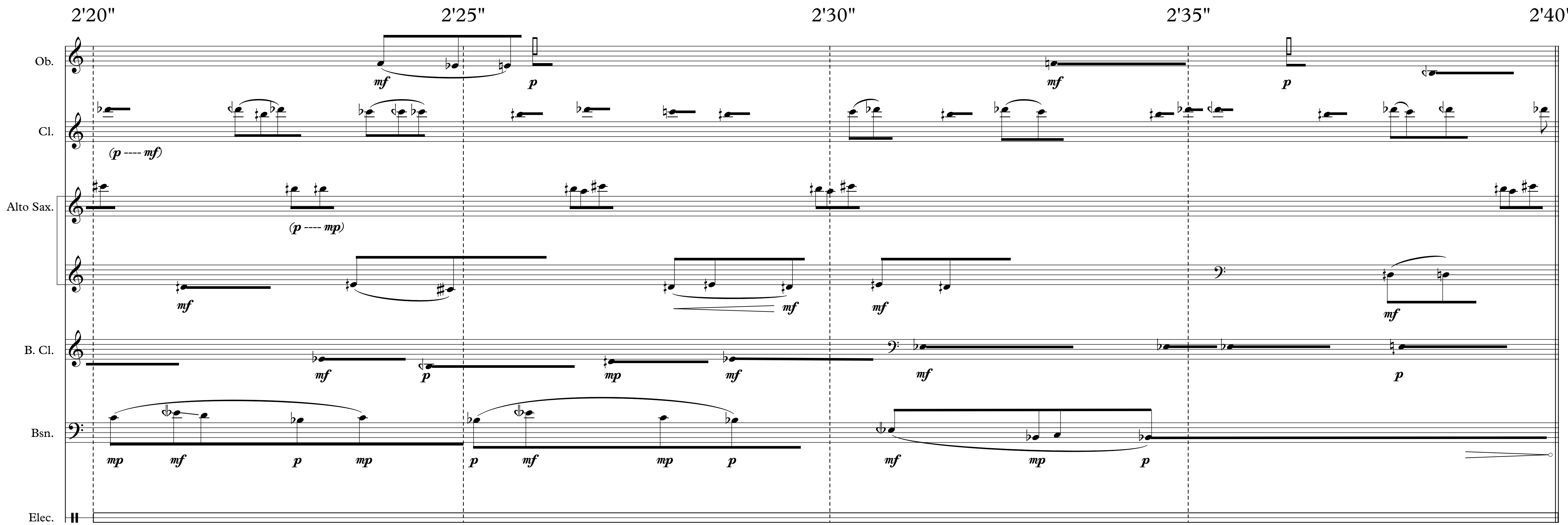

1'
1'05"
1'10"
1'15"
1'20"
Ob.
(pp ---- mp)
p
Cl.
(pp ---- mp)
(p)
Alto Sax.
(pp ---- mp)
(p)
IR
B. Cl.
(pp ---- mp)
f
Bsn.
(p)
(mp)
mf
attacca

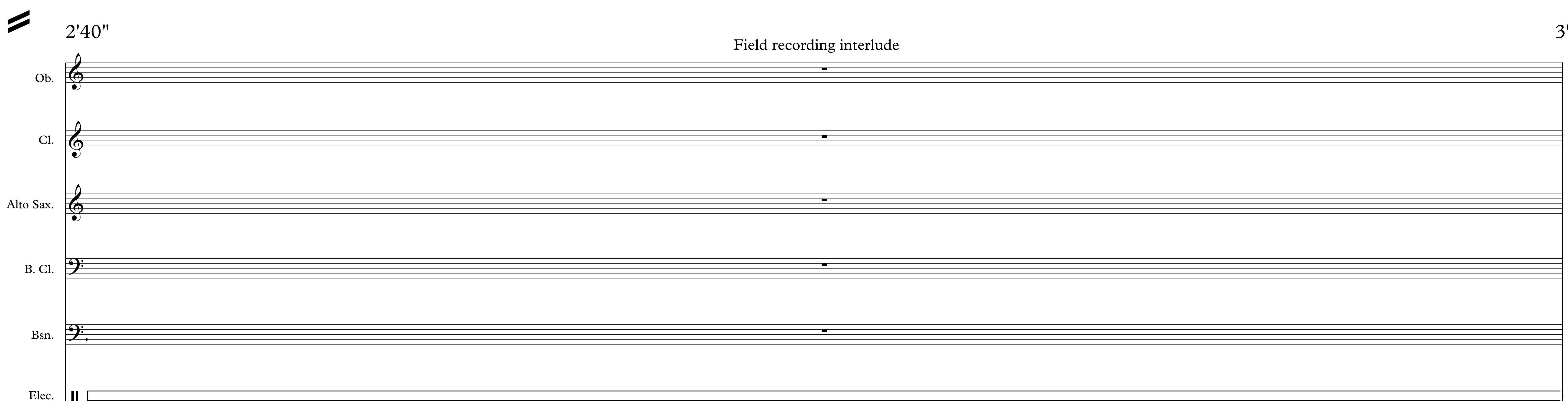
2.

*) Noise (see "imitative noise" in performance notes)

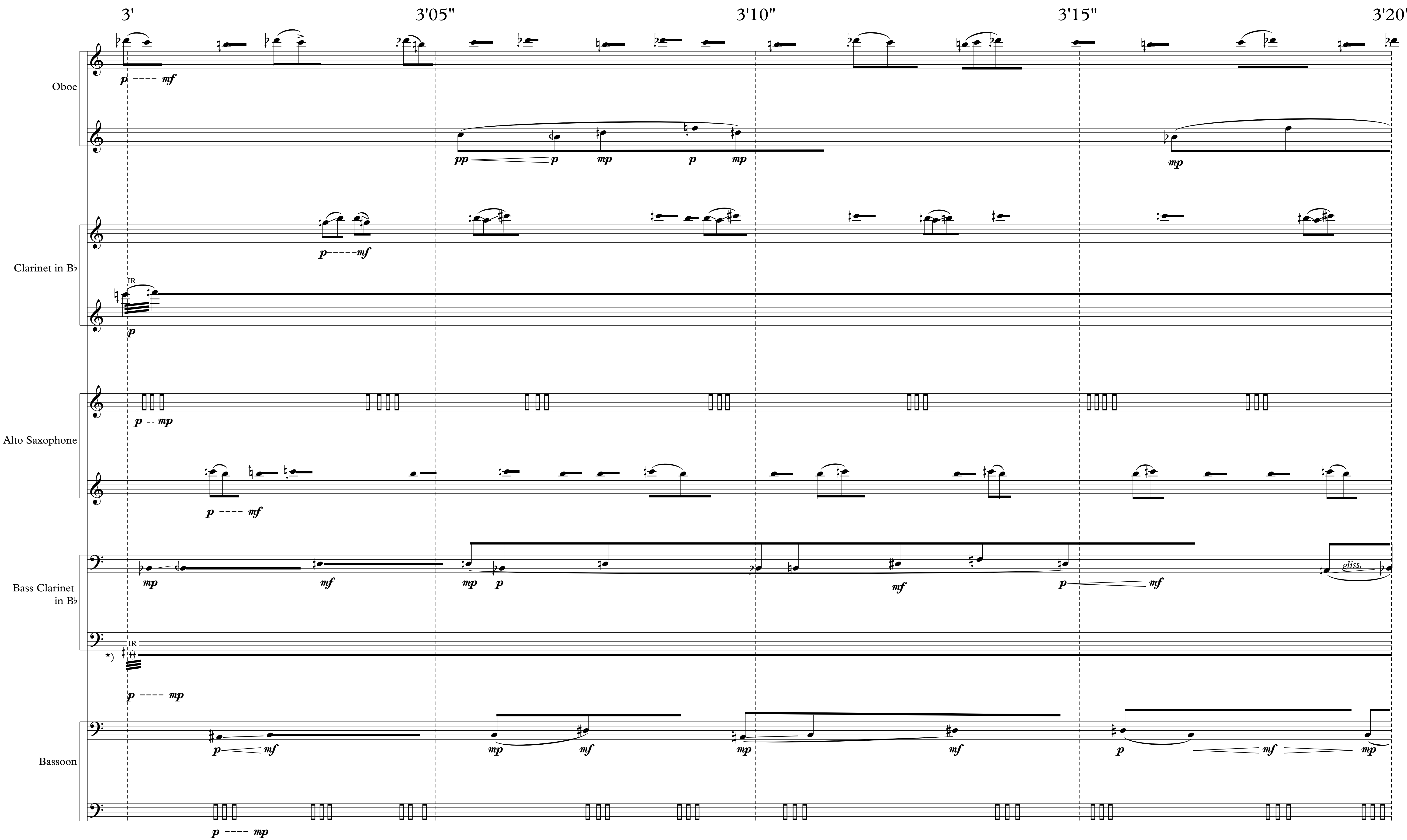



2'
2'05"
2'10"
2'15"
2'20"
Ob.
Cl.
Alto Sax.
B. Cl.
Bsn.
Elec.
mp
p
mp
p
p
mp
p
mp
p
(p ---- mf)
(p ---- mp)
mf
mp
p
mp
p
mf
mp
mf
mf
p
mf
Spatialized field recording: section 2
mp

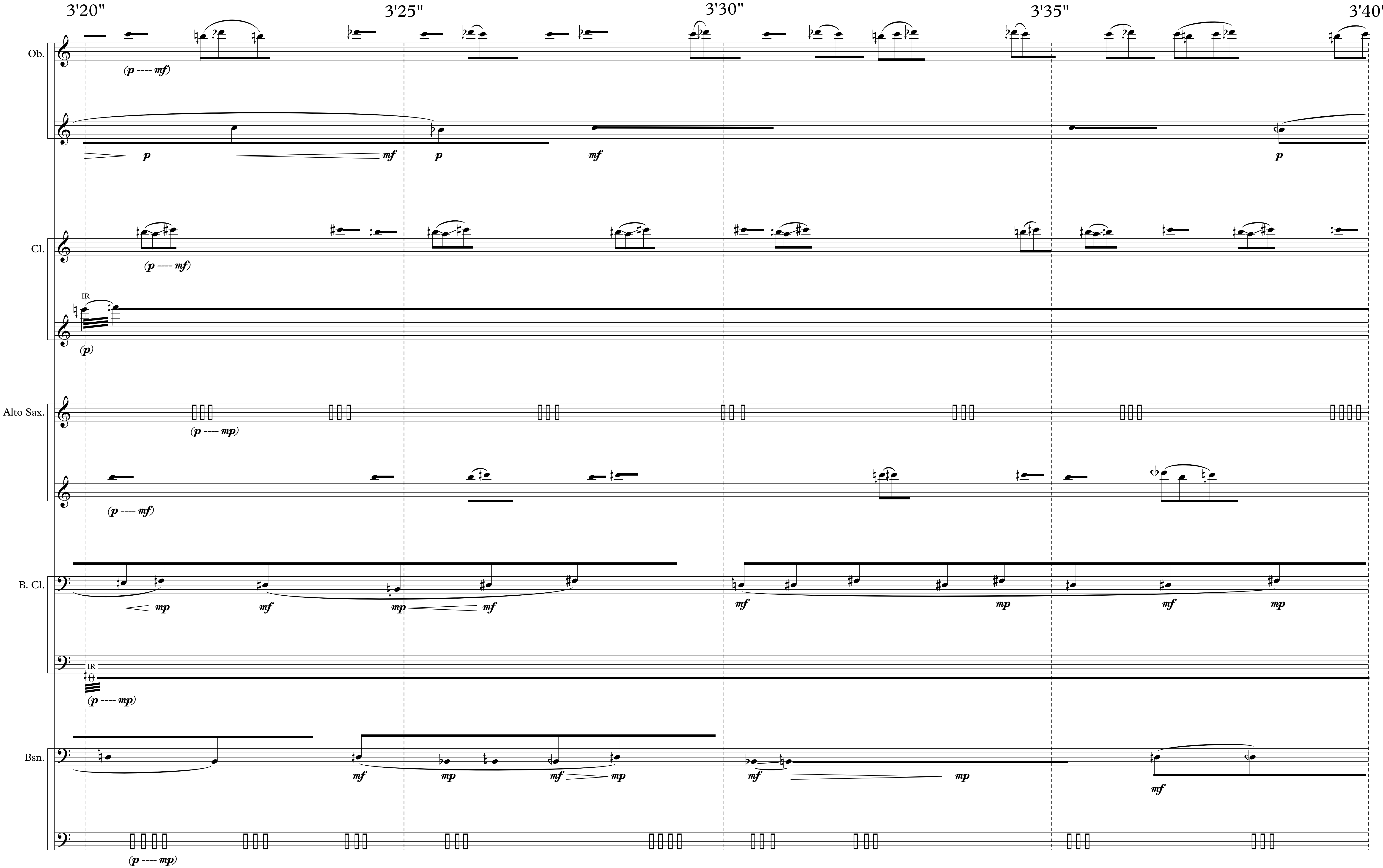

2'20"
2'25"
2'30"
2'35"
2'40"
Ob.
Cl.
Alto Sax.
B. Cl.
Bsn.
Elec.
mf
p
(p ---- mf)
(p ---- mp)
mp

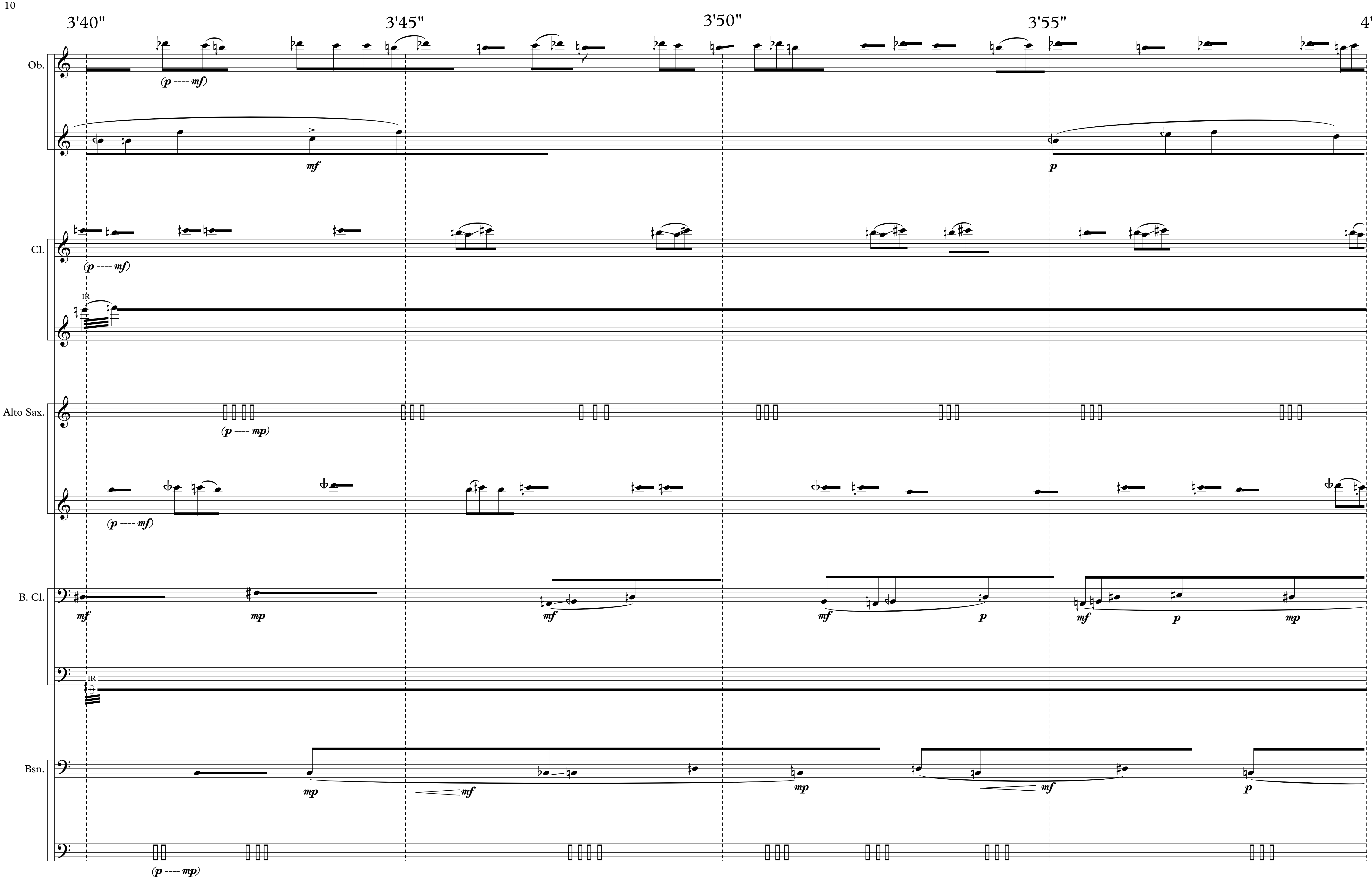

2'40"
Field recording interlude
3'
Ob.
Cl.
Alto Sax.
B. Cl.
Bsn.
Elec.

3.

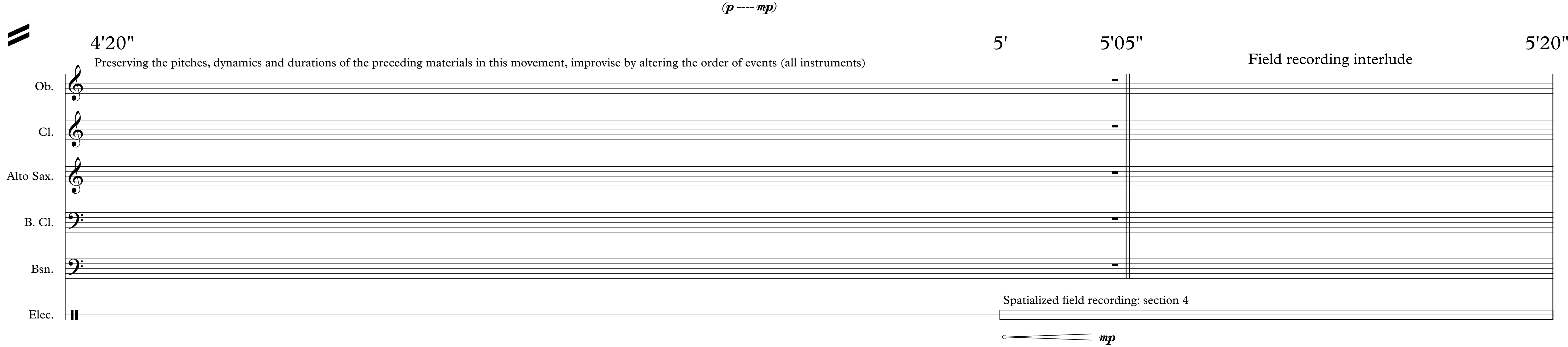


*) Include all the possible pitches within the given range in tremolo with irregular durations and patterns.

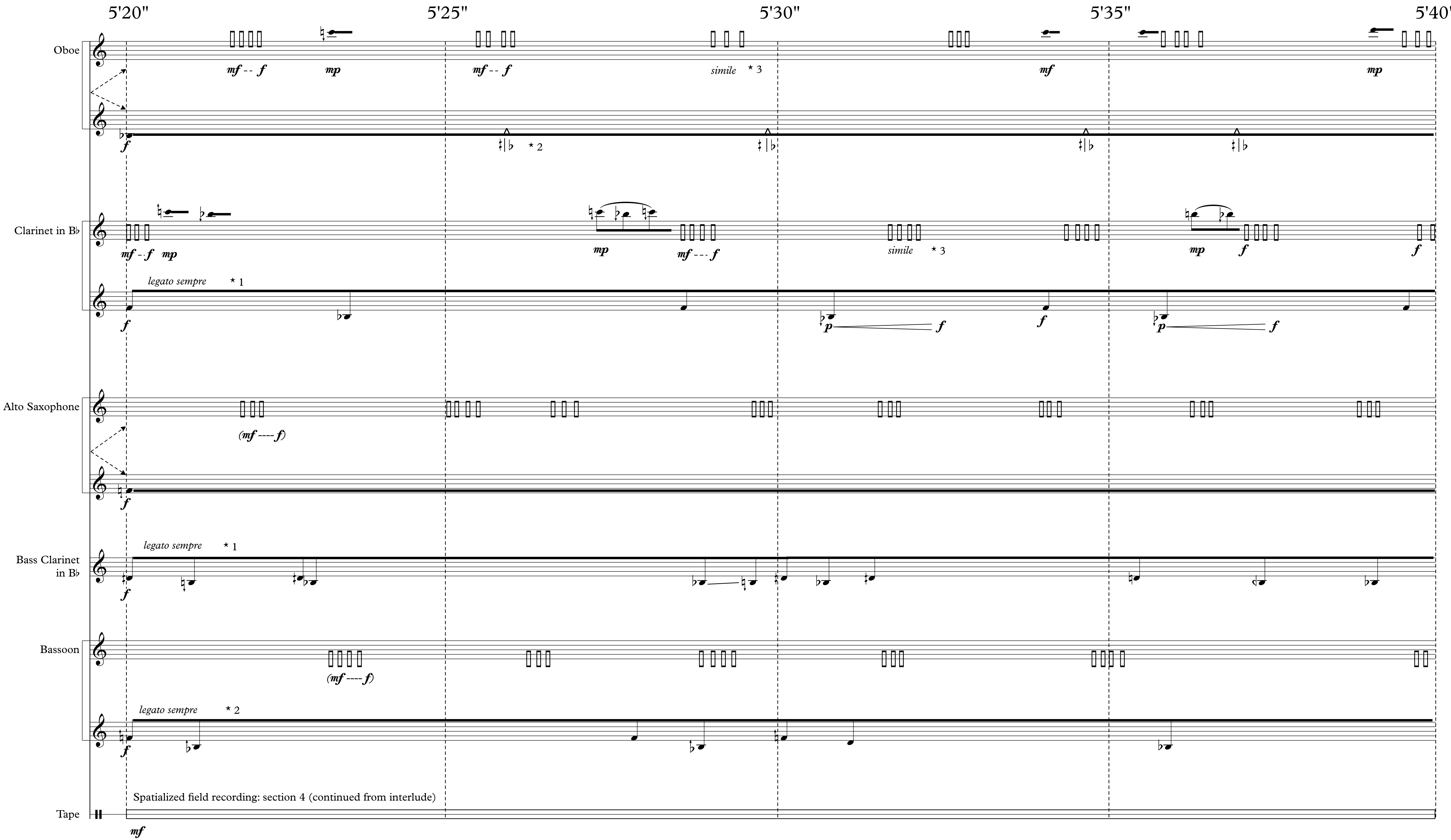

3'20"
3'25"
3'30"
3'35"
3'40"
Ob.
(p ---- mf)
p
mf
p
mf
p
Cl.
(p ---- mf)
IR
(p)
Alto Sax.
(p ---- mp)
(p ---- mf)
B. Cl.
mp
mf
mp
mf
mf
mp
mf
mp
IR
(p ---- mp)
Bsn.
mf
mp
mf
mp
mf
mp
mf
(p ---- mp)


3'40"
3'45"
3'50"
3'55"
4'
Ob.
(p ---- mf)
mf
p
Cl.
(p ---- mf)
IR
Alto Sax.
(p ---- mp)
(p ---- mf)
B. Cl.
mf
mp
mf
mf
p
mf
p
mp
IR
Bsn.
mp
mf
mp
mf
p
(p ---- mp)


4'
4'05"
4'10"
4'15"
4'20"
Ob.
(p ---- mf)
mf
mf
Cl.
(p ---- mf)
IR
Alto Sax.
(p ---- mp)
(p ---- mf)
B. Cl.
mf
IR
Bsn.
p
mp
mf
(p ---- mp)


4'20"
5'
5'05"
5'20"
Preserving the pitches, dynamics and durations of the preceding materials in this movement, improvise by altering the order of events (all instruments)
Field recording interlude
Ob.
Cl.
Alto Sax.
B. Cl.
Bsn.
Elec.
Spatialized field recording: section 4
mp

4.

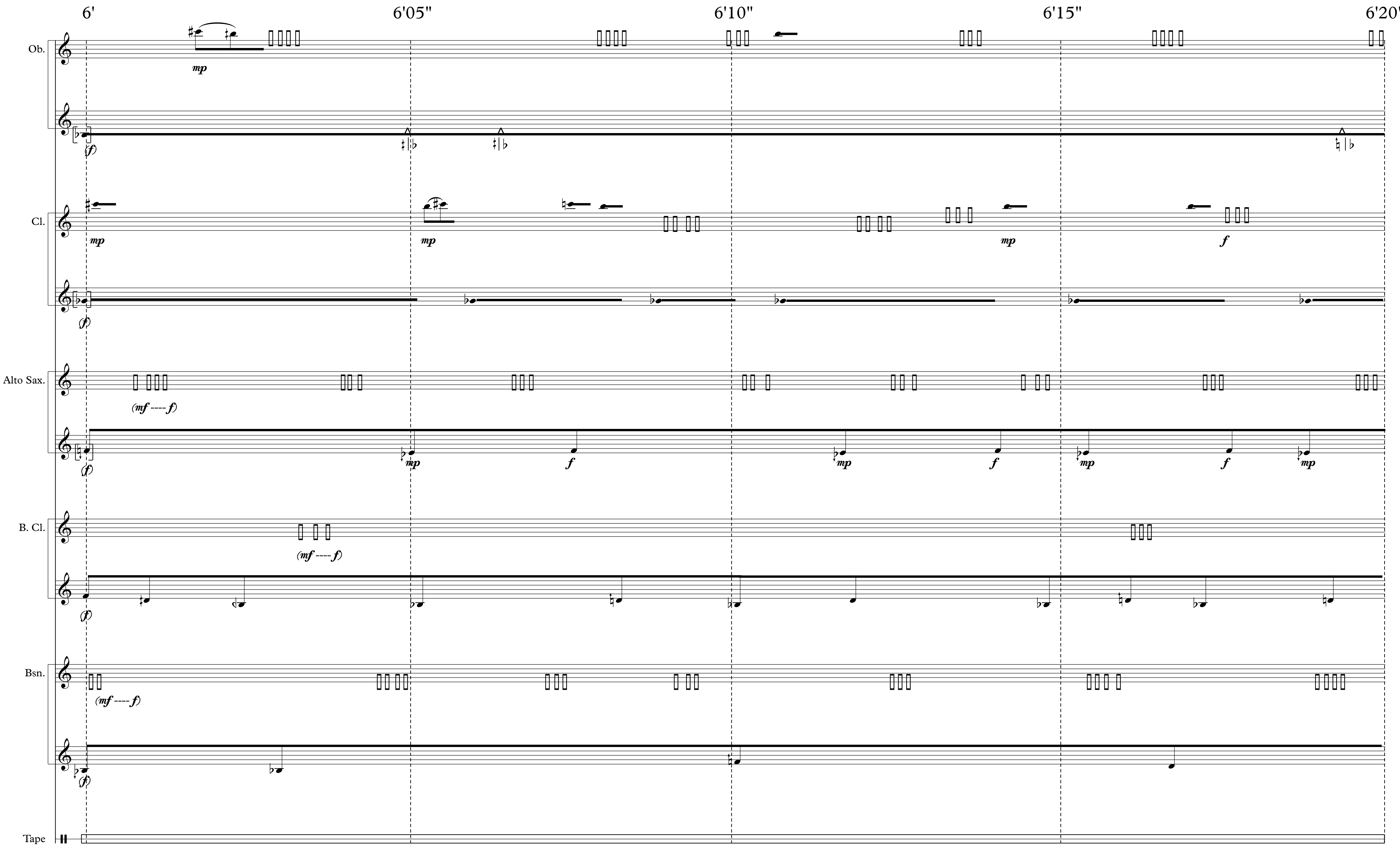


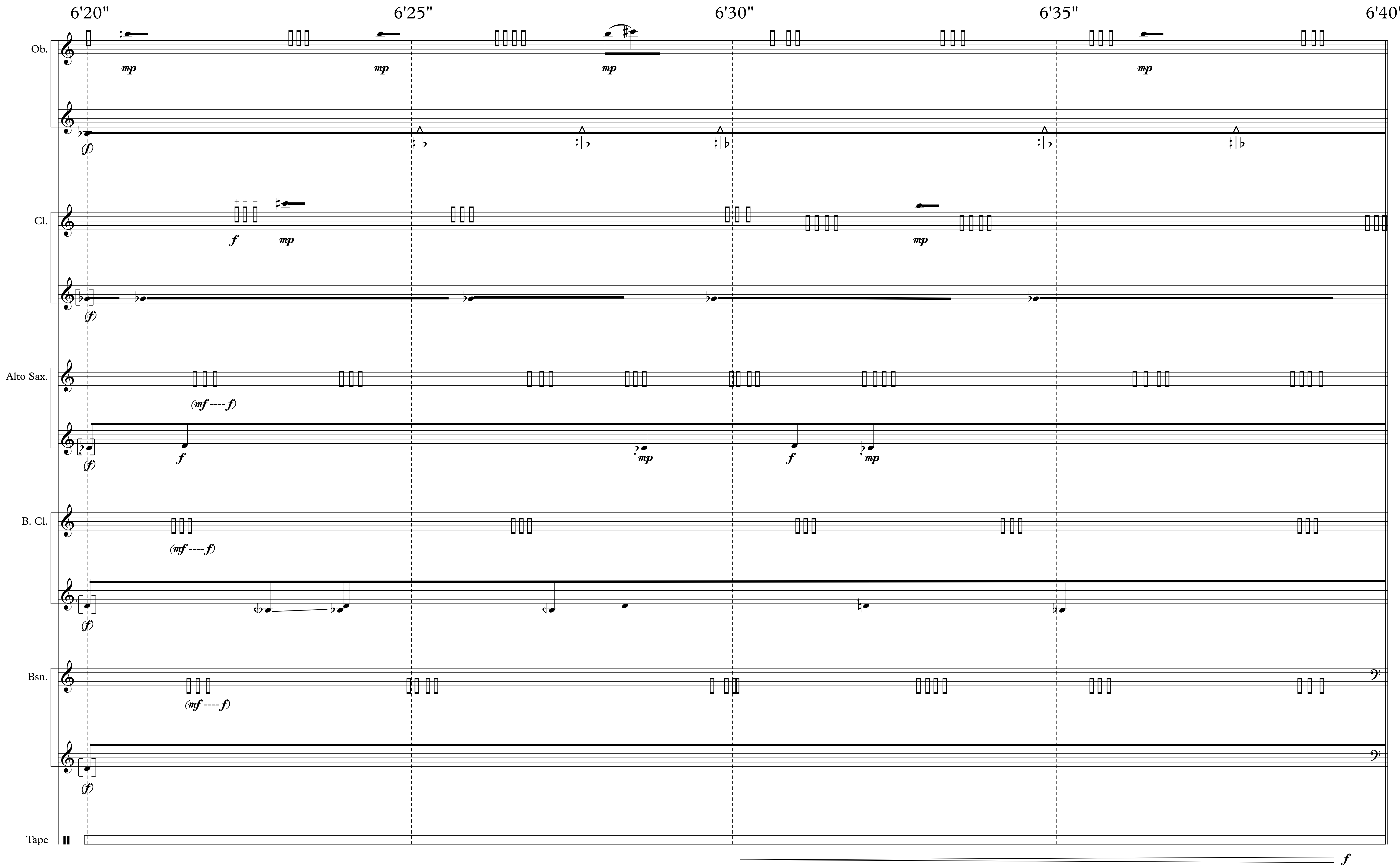
* 1) Breathe when needed, ad. lib

* 2) Very brief inflection to the accidental on the left side of the vertical slash

* 3) The noise material (rectangle noteheads) should always present different dynamics, based on the given range in the first bar. This dynamics procedure does not apply to other material (standard noteheads) in the staff.

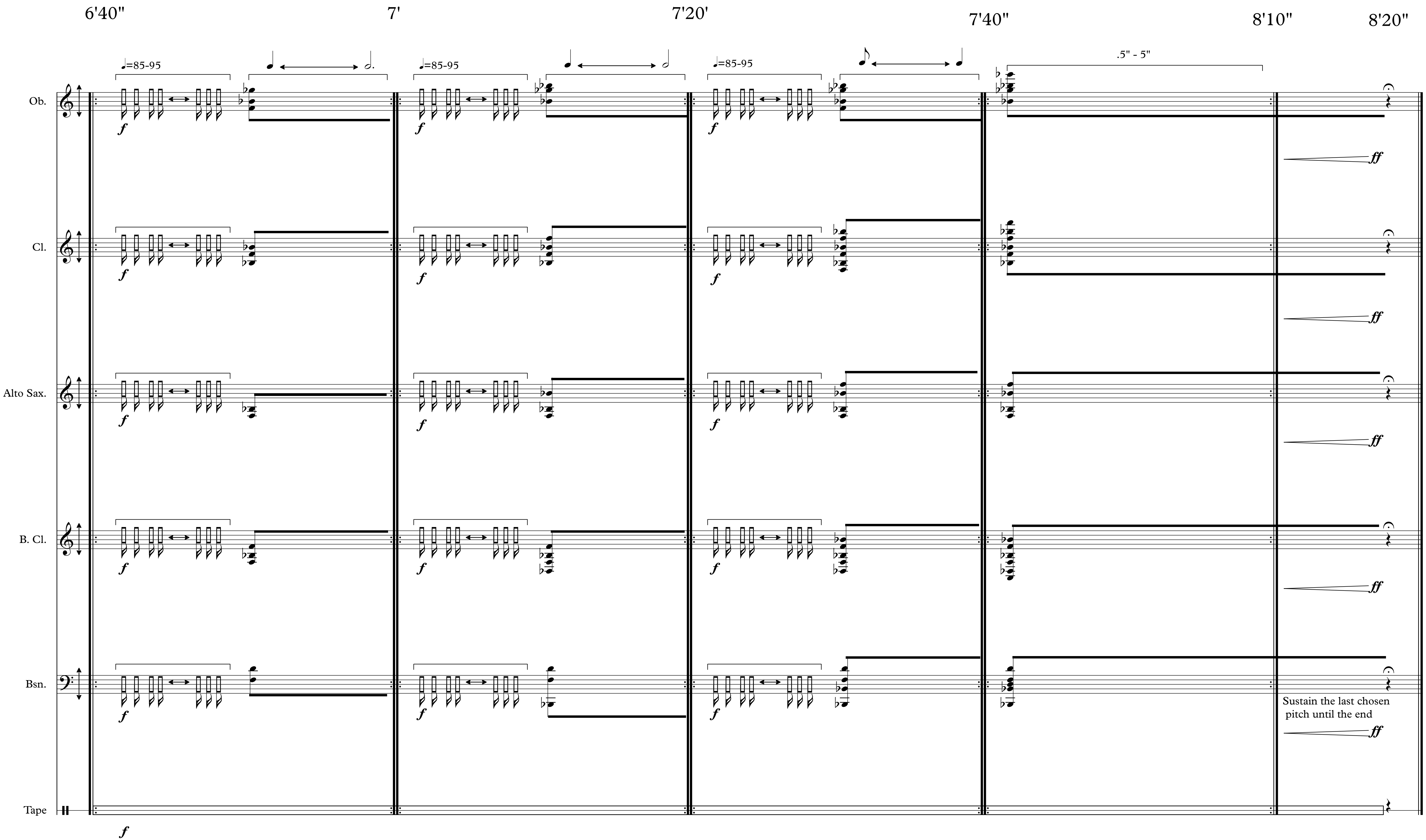

5'40"
5'45"
5'50"
5'55"
6'
Ob.
mp
mp
mp
mp
(f)
Cl.
mp
mp
mp
f
p
f
p
f
p
mf
f
Alto Sax.
(mf ---- f)
(f)
B. Cl.
(mf ---- f)
(f)
Bsn.
(mf ---- f)
(f)
Tape


6'
6'05"
6'10"
6'15"
6'20"
Ob.
mp
(f)
Cl.
mp
mp
mp
f
(f)
Alto Sax.
(mf ---- f)
(f)
mp
f
mp
f
mp
f
mp
B. Cl.
(mf ---- f)
(f)
Bsn.
(mf ---- f)
(f)
Tape


6'20"
6'25"
6'30"
6'35"
6'40"
Ob.
mp
mp
mp
mp
(f)
Cl.
f
mp
mp
(f)
Alto Sax.
(mf ---- f)
(f)
f
mp
f
mp
B. Cl.
(mf ---- f)
(f)
Bsn.
(mf ---- f)
(f)
Tape
f

attacca

5.

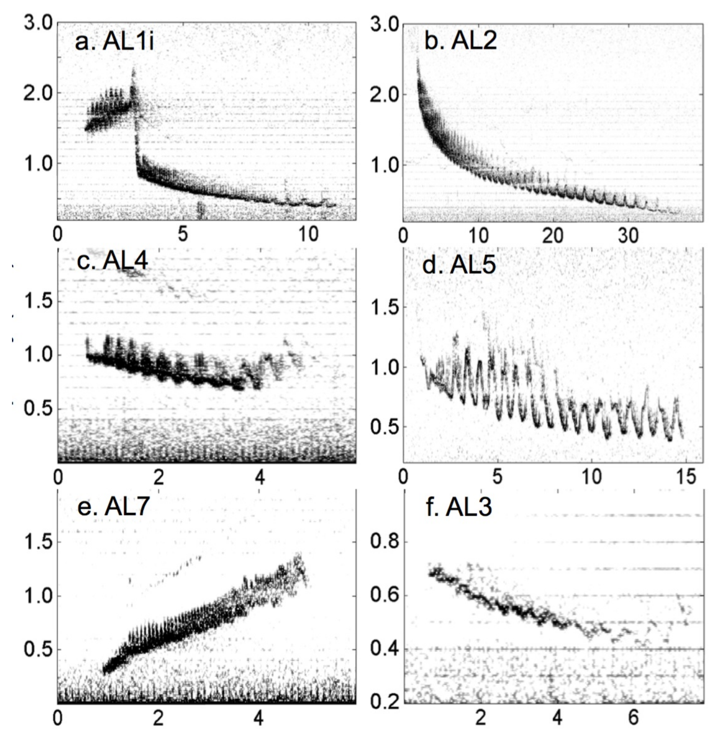


*See performance notes, ("Indeterminate material cells").

Under the sea ice

for string quartet and electronics

Meditations on the songs of the Arctic bearded seals



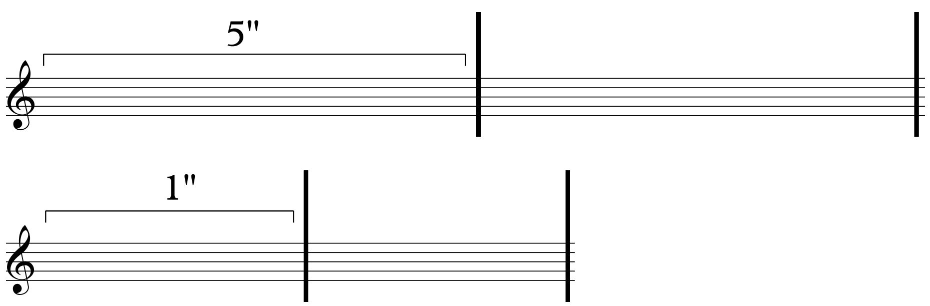
Christopher Luna-Mega

Special thanks to Joshua Jones, Staff Research Associate
at the Scripps Whale Acoustic Lab (University of California, San Diego)
for providing the recordings and information that made this piece possible

Performance Notes

Proportional Notation

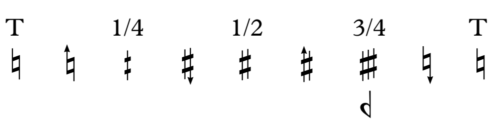
Each bar in the score is either 1” or 5” long.
Brackets announce a change in the duration of a bar, which will affect the subsequent bars until a new change occurs.



Durations / Rests

Stems do not indicate duration –their purpose is visual reference.
Sounds must be continued until followed by another sound or by a silence.

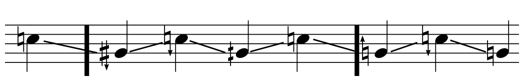
Accidentals



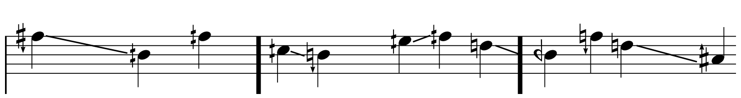
In various points of the piece, especially the fast sections, playing approximately the suggested accidental should suffice.

Pitch references in glissandi

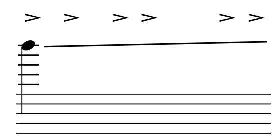
Note heads preceded and succeeded by glissandi must not be attacked. They serve as precise pitch references.



Pitches without a preceding glissando must be attacked



Bow pressure accents over glissando

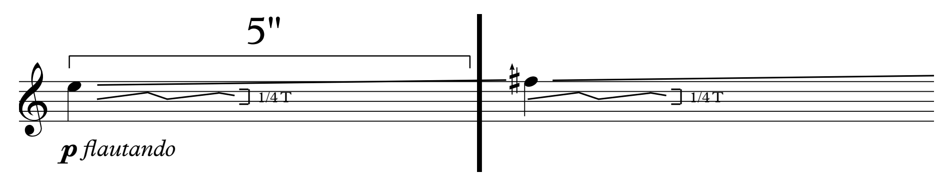


Fast irregular increase in bow pressure while glissando

Tremolo

Tremolos should be played as fast as possible.
Occasionally perform irregular rhythms (ad. libitum).

Ad libitum glissandi



Ad libitum gradual fluctuations around the notated pitches (i.e. ¼ of a tone higher and ¼ tone lower than E) and dynamics (i.e. *pp* and *mp*). The fraction in the right side of the bracket indicates the intervallic frame for the ad lib. glissandi (i.e. ¼ of a tone or ½ of a tone around the written note).

The ad. lib. glissando figure lasts the full bar in which it is notated. The pitches in between the figure must not be attacked (they are referential).

Time cues and synchronization

Time cues are provided consistently throughout the score. Players must be synchronized to these cues due to their interaction with the fixed media. This may be achieved by one or more of these options: 1. Using individual timers on the stand; 2. Using click tracks; 3. A conductor.

Electronics

An audio file (stereo) with the tape part for the piece is provided electronically. Fade-ins and fade-outs indicated in the score have been previously set in the audio file. The track must be leveled and pre-set in dress rehearsal prior to a performance. Please e-mail ch.luna.mega@gmail.com to request the audio file.

The output devices used for playback are flexible, from 2 speakers on stage to several speakers surrounding the audience, depending on the technology available.

Supplementary material

All the musical materials performed by the string quartet derive from direct transcriptions and arrangements of underwater songs of bearded seals in the Chukchi Sea, north of Point Barrow, Alaska.

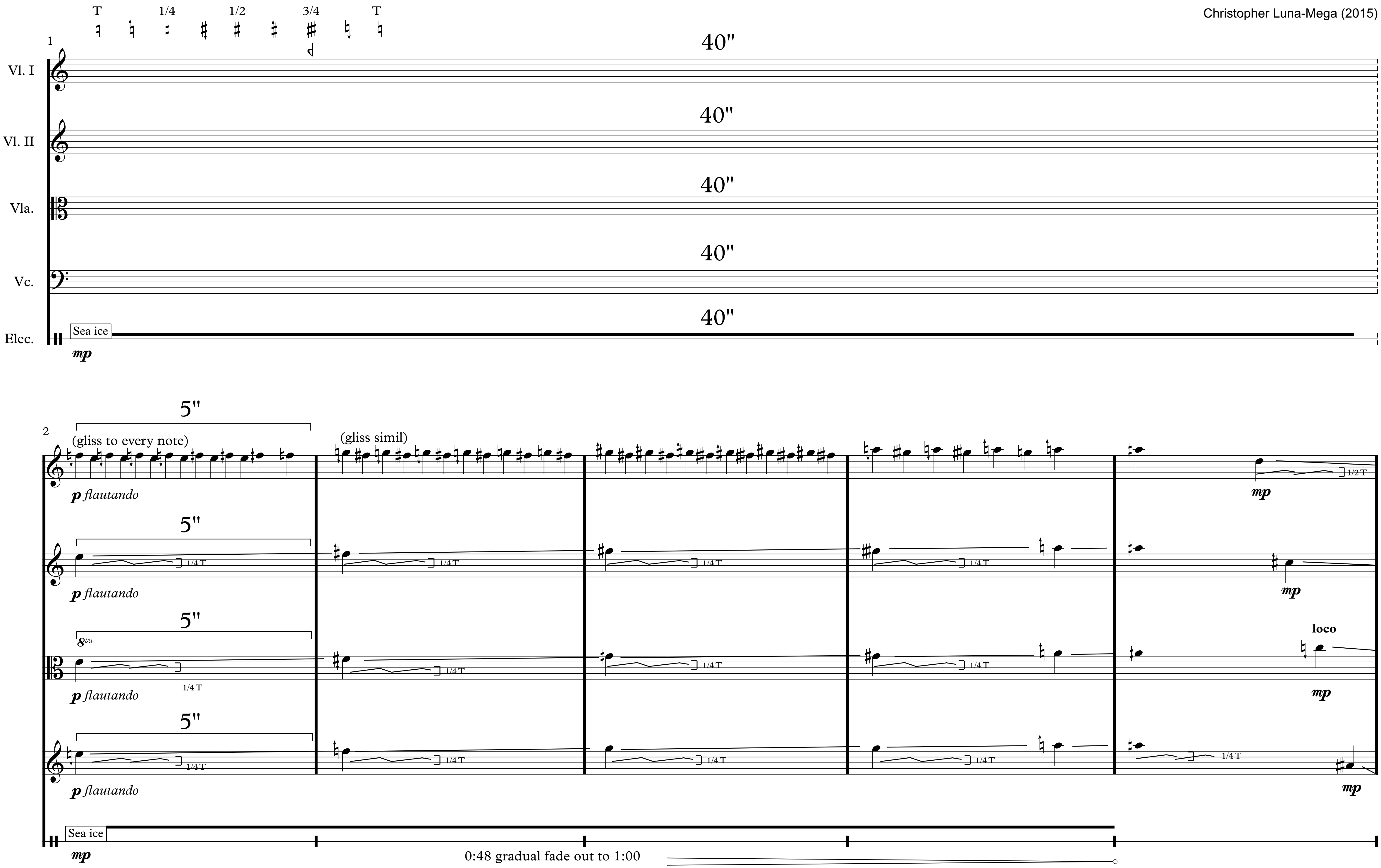
Included in the parts are the transcriptions of each of the seal songs, from which all the materials in the piece are developed.

Together with the audio file to be played back in the performance of the piece, there is a folder with audio files for each seal call, with labels that correspond to those of the transcriptions included in the parts. Please take a few moments prior to rehearsal to play these audio while looking at the corresponding transcriptions. Listening to the source material is the essence of this work.

Under the sea ice

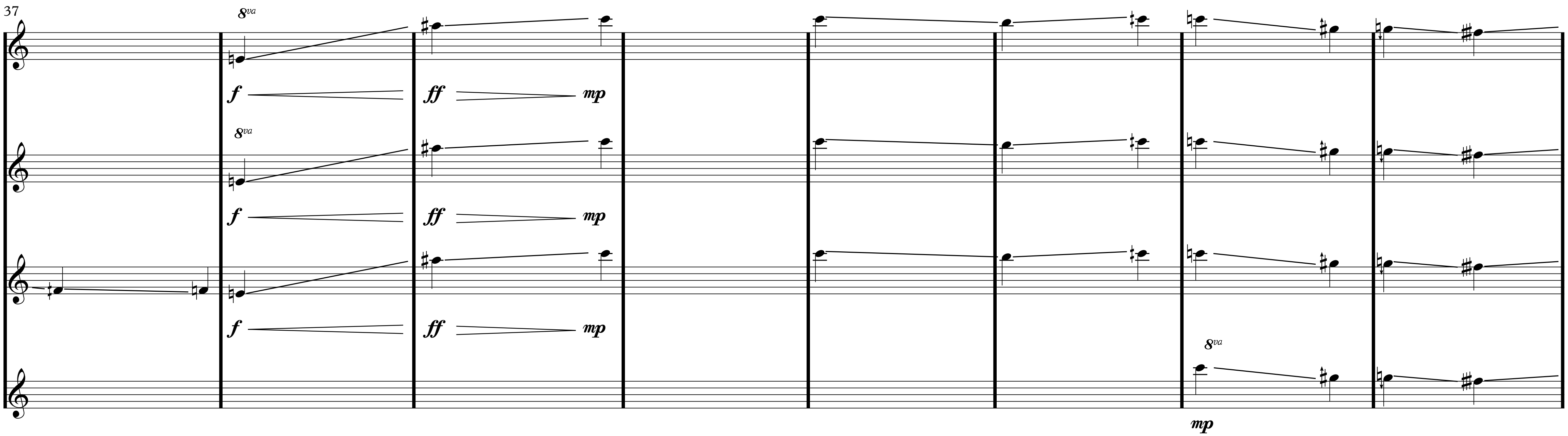
for string quartet and electronics

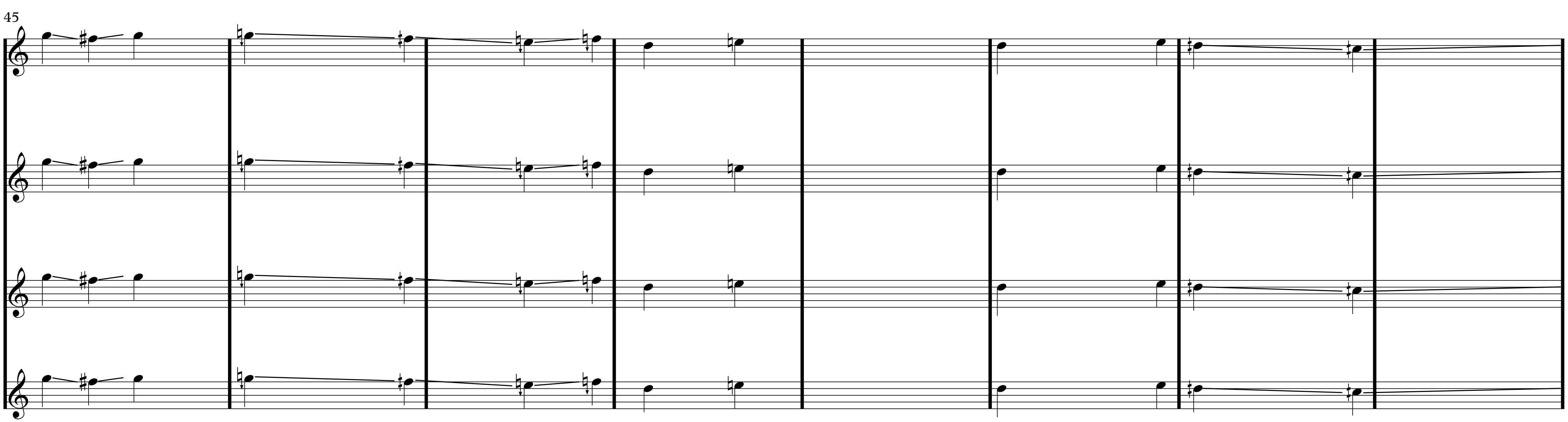
Christopher Luna-Mega (2015)



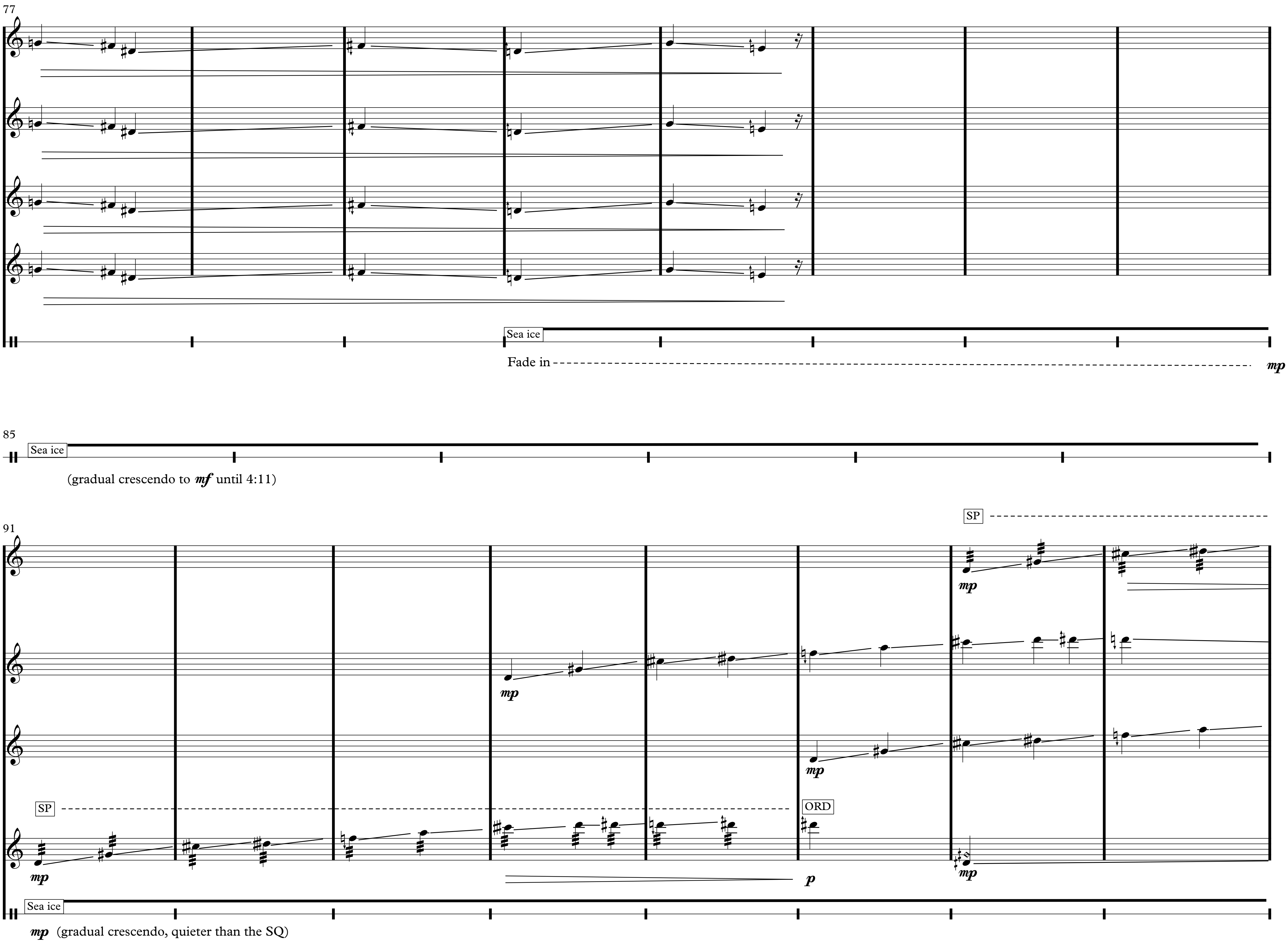

7
1/2 T
12
mf
p
Sea ice
gradual fade in/crescendo to mf until 2:10

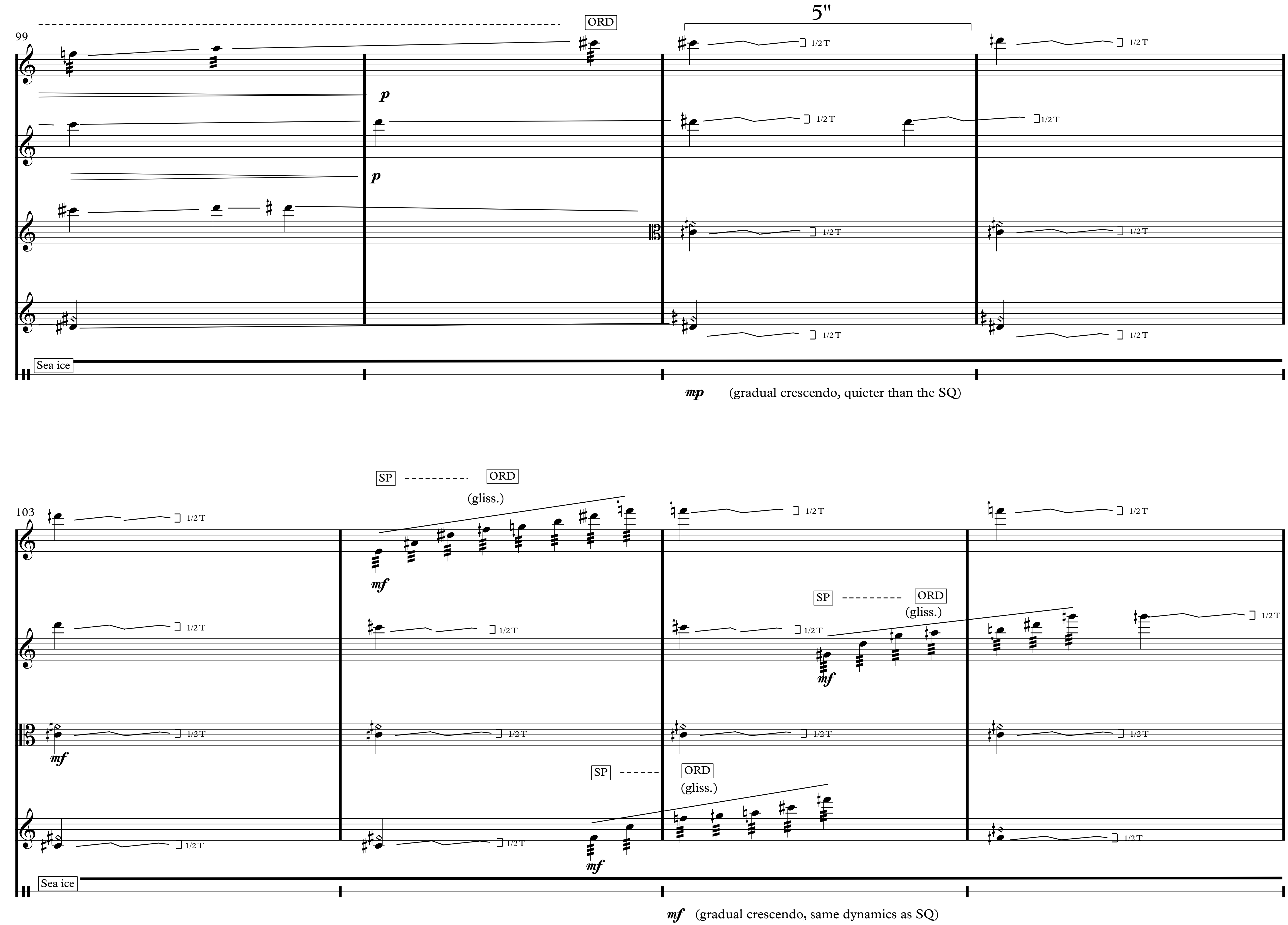

17
1/2 T
1"
mf
f
21
8va
p
mf
p
f
f
29
f
f
f
f
f
p
f
p
mf
p

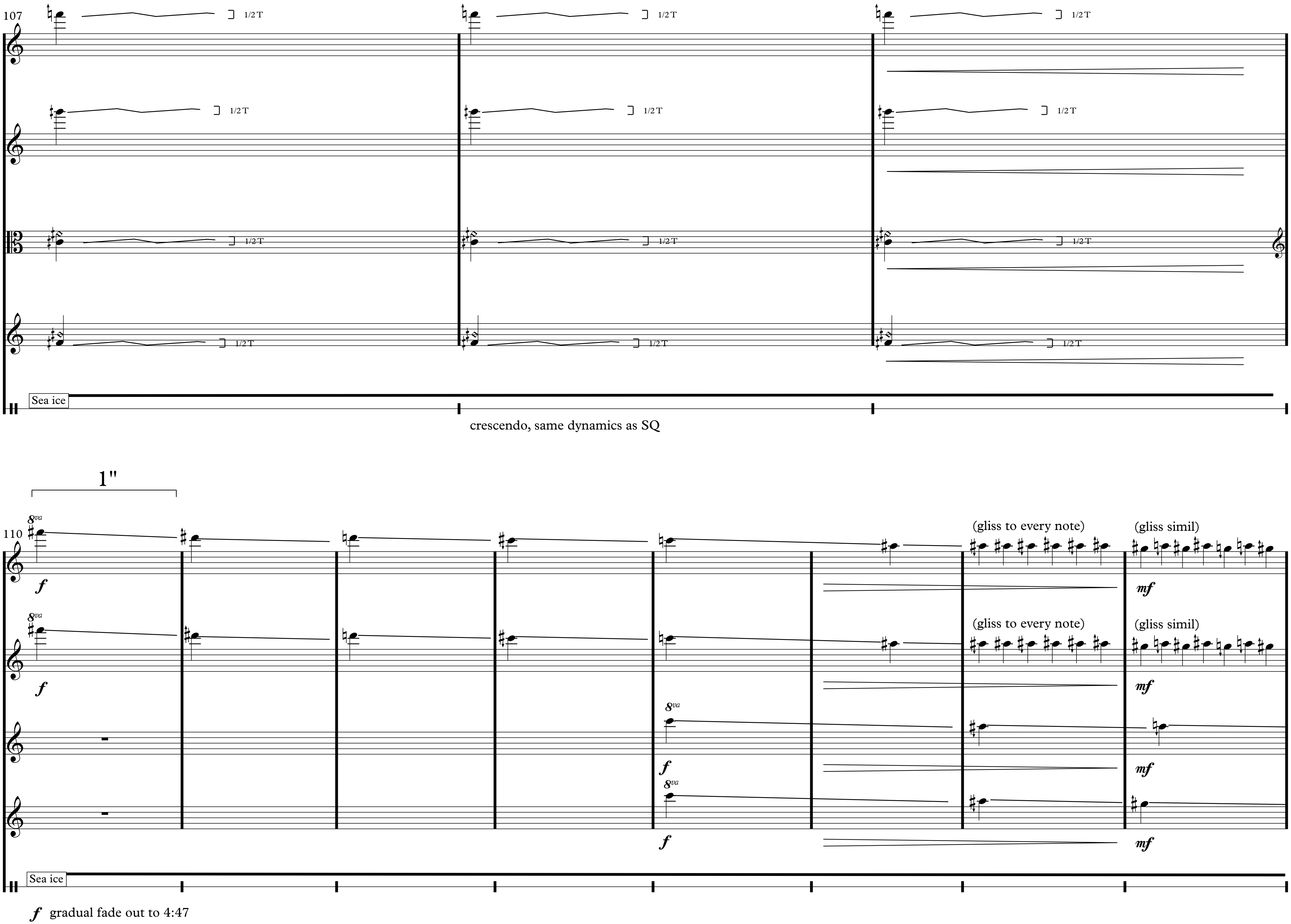

37
8va
f
ff
mp
8va
f
ff
mp
f
ff
mp
8va
mp

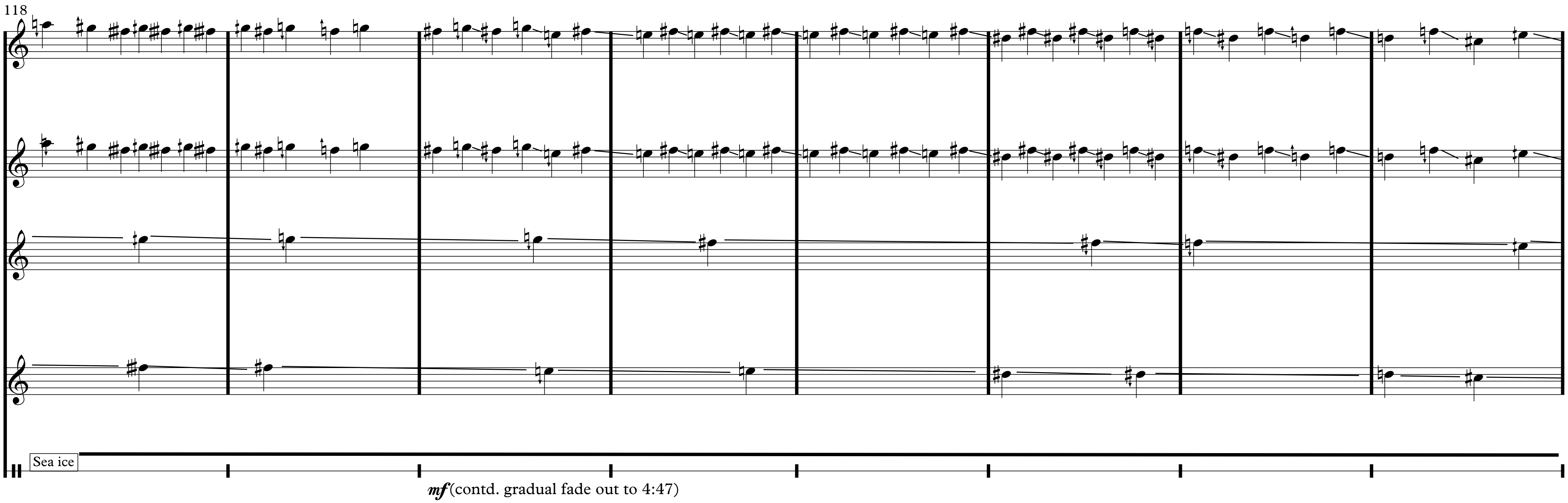

45

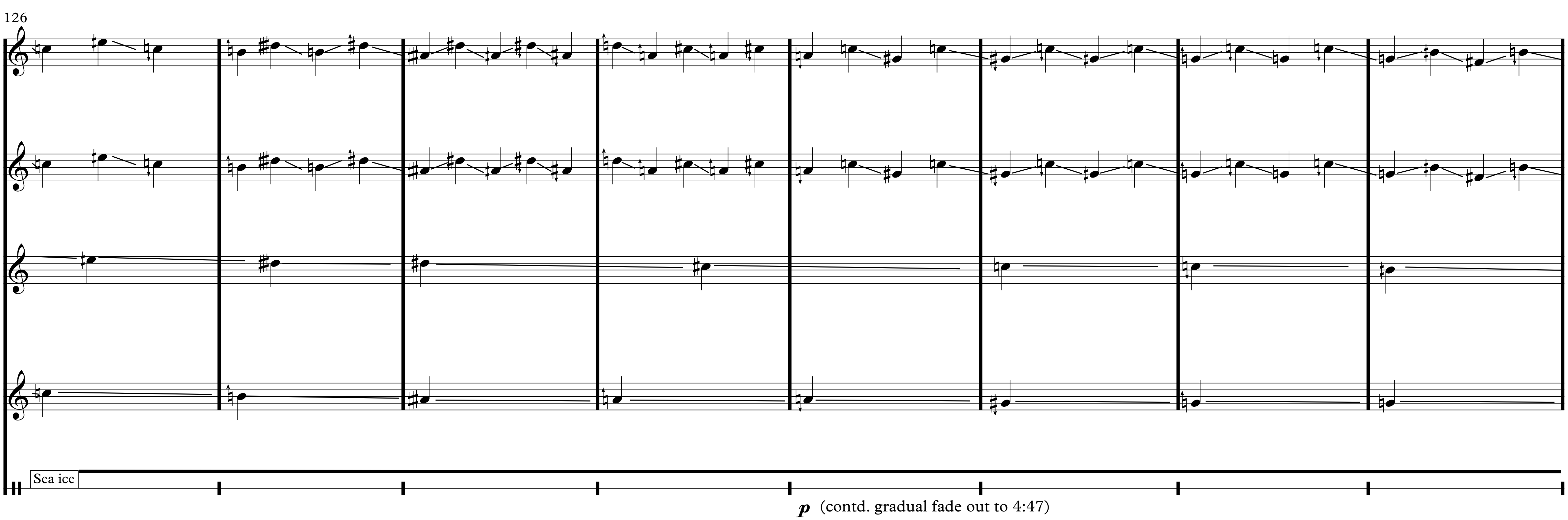

53
61
loco
loco
loco
loco
69

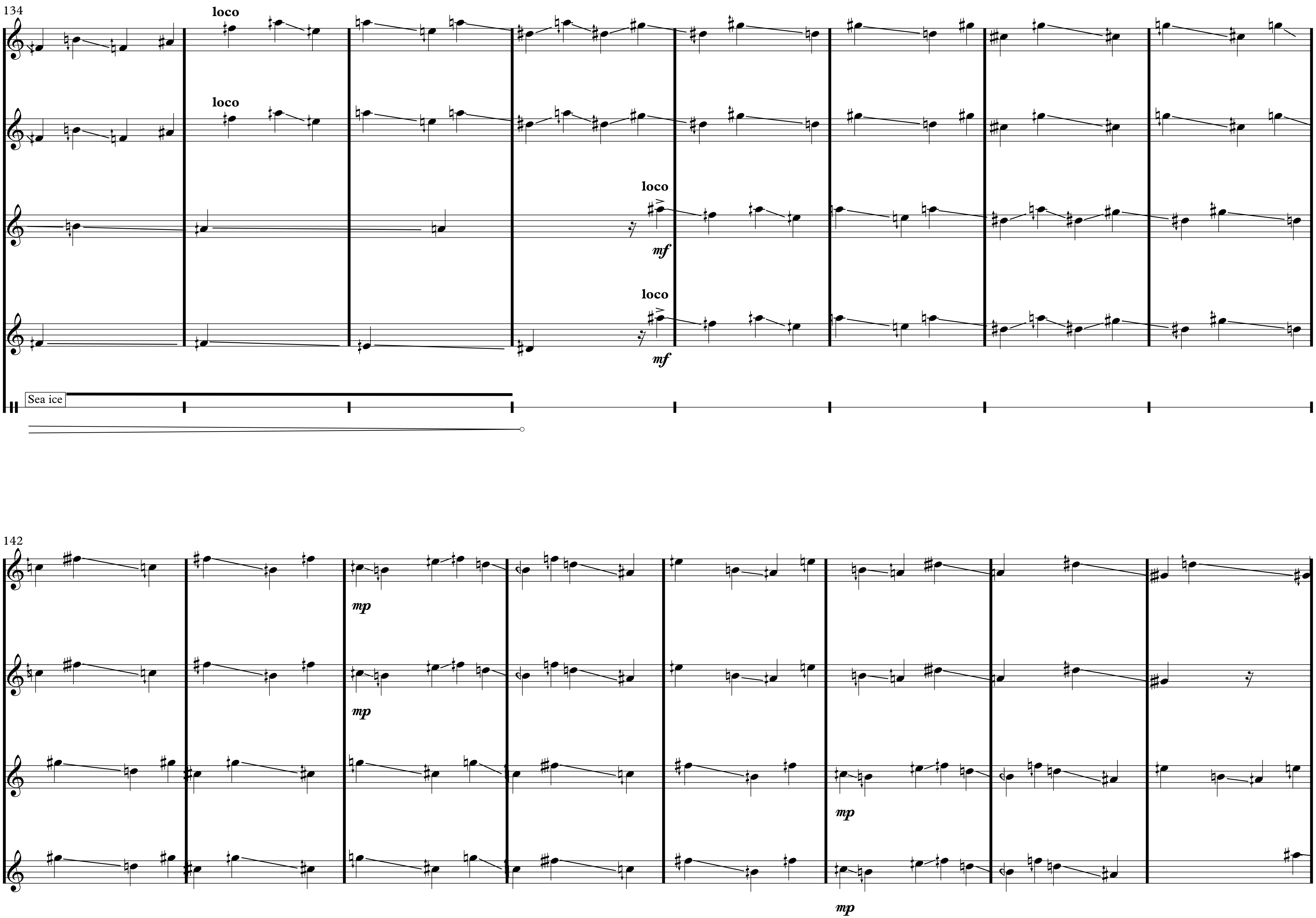

77
Sea ice
Fade in
mp
85
Sea ice
(gradual crescendo to mf until 4:11)
91
SP
SP
mp
mp
mp
ORD
p
mp
Sea ice
mp (gradual crescendo, quieter than the SQ)

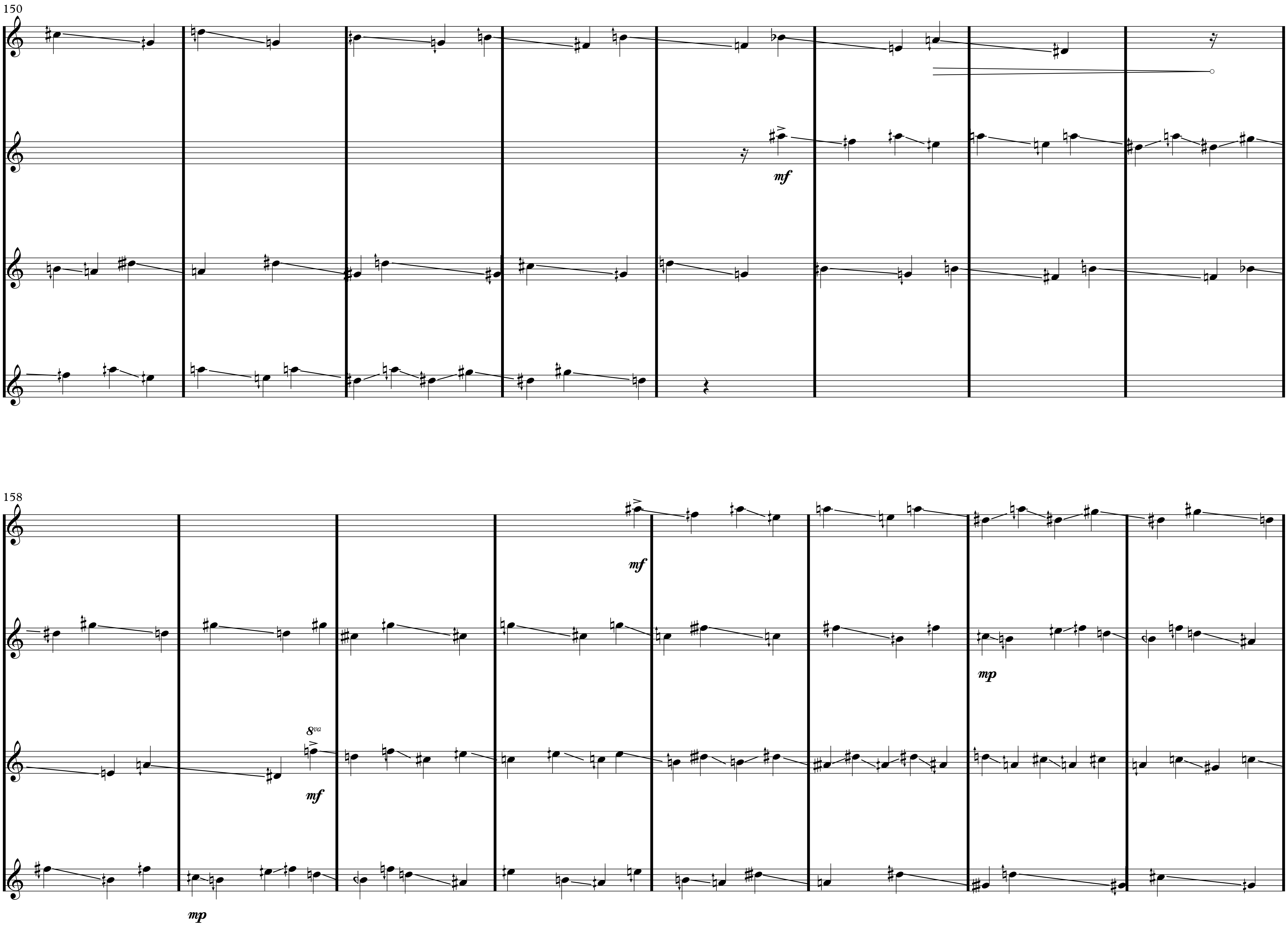

99
ORD
5"
1/2T
p
p
Sea ice
mp (gradual crescendo, quieter than the SQ)
103
SP
ORD
(gliss.)
mf
1/2T
SP
ORD
(gliss.)
mf
mf
SP
ORD
(gliss.)
mf
Sea ice
mf (gradual crescendo, same dynamics as SQ)

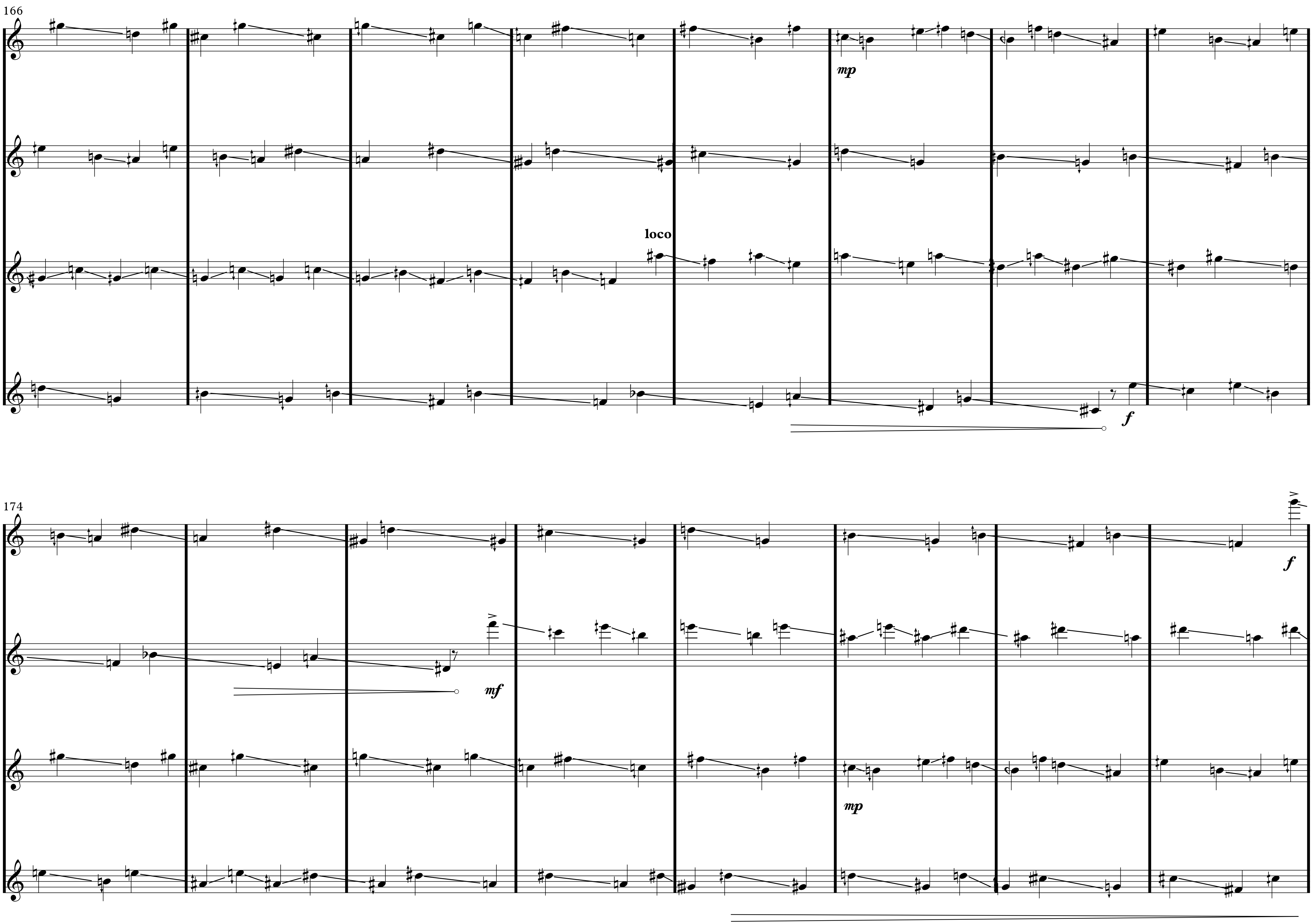

107
1/2 T
Sea ice
crescendo, same dynamics as SQ
1"
110
8va
f
(gliss to every note)
(gliss simil)
mf
Sea ice
f gradual fade out to 4:47

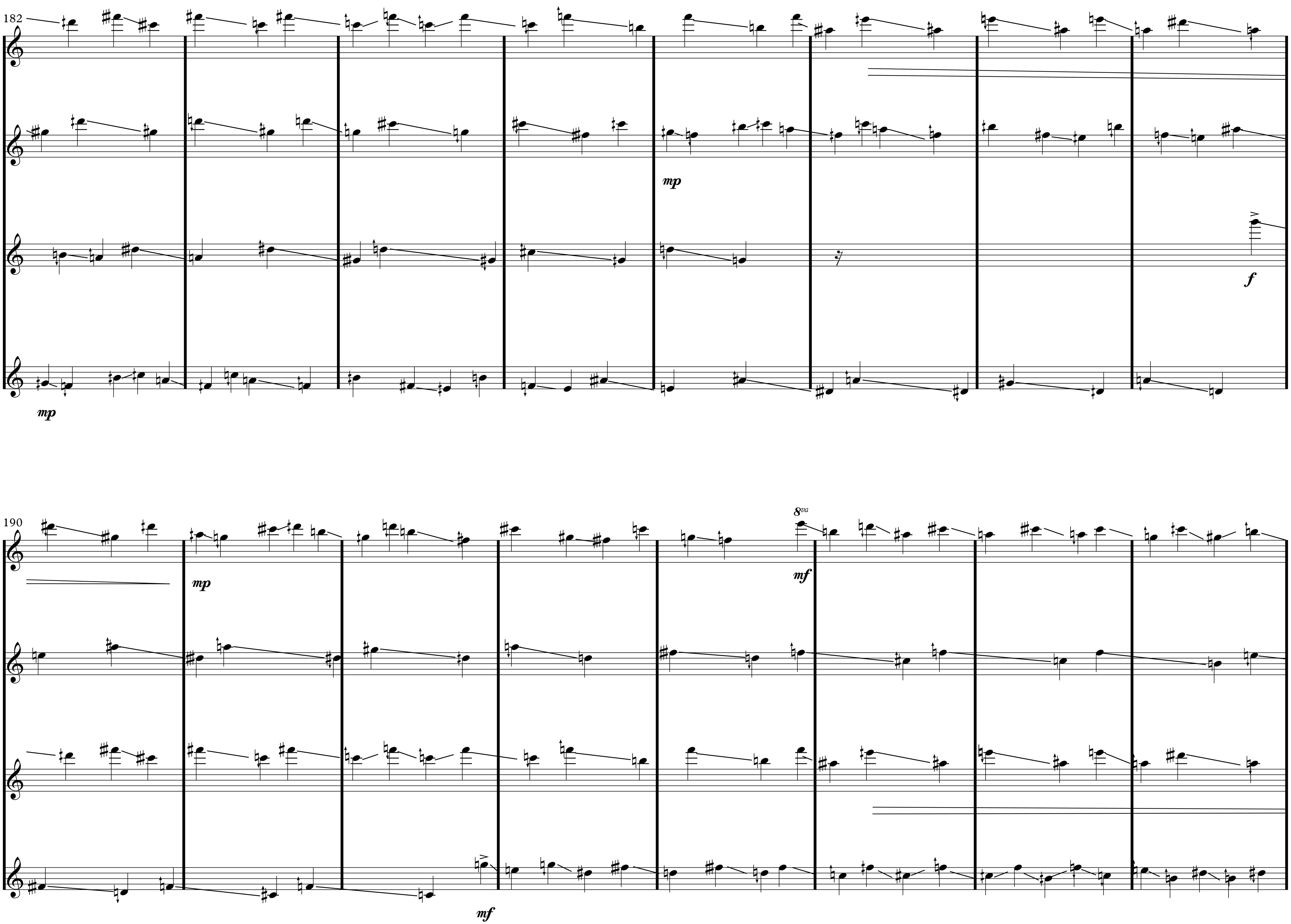

118
Sea ice
mf (contd. gradual fade out to 4:47)

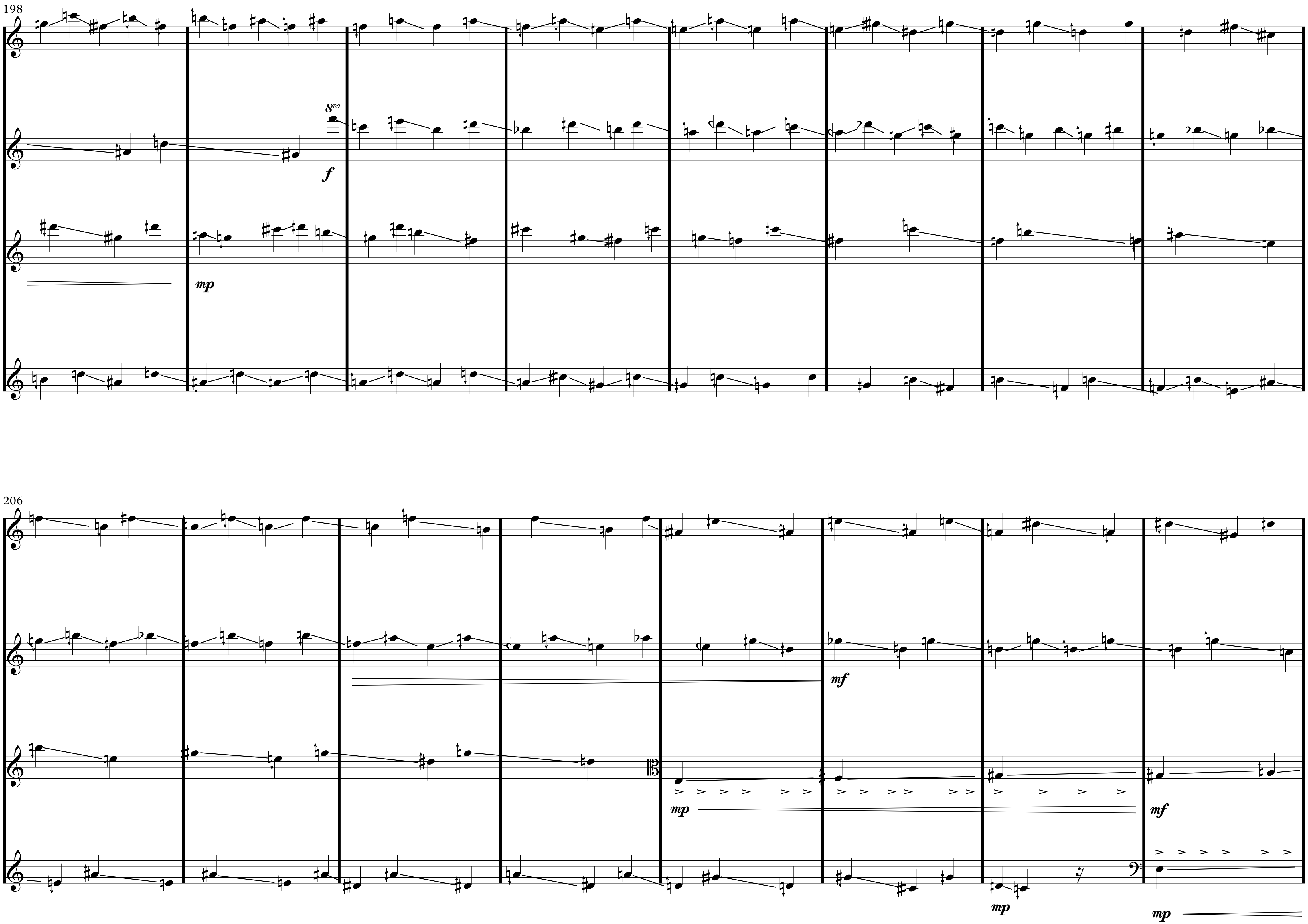

126
Sea ice
p (contd. gradual fade out to 4:47)


134
loco
loco
loco
mf
loco
mf
Sea ice
142
mp
mp
mp
mp


150
mf
158
mf
8va
mf
mp
mp


166
mp
loco
f
174
f
mf
mp


182
mp
mp
f
190
mp
8va
mf
mf

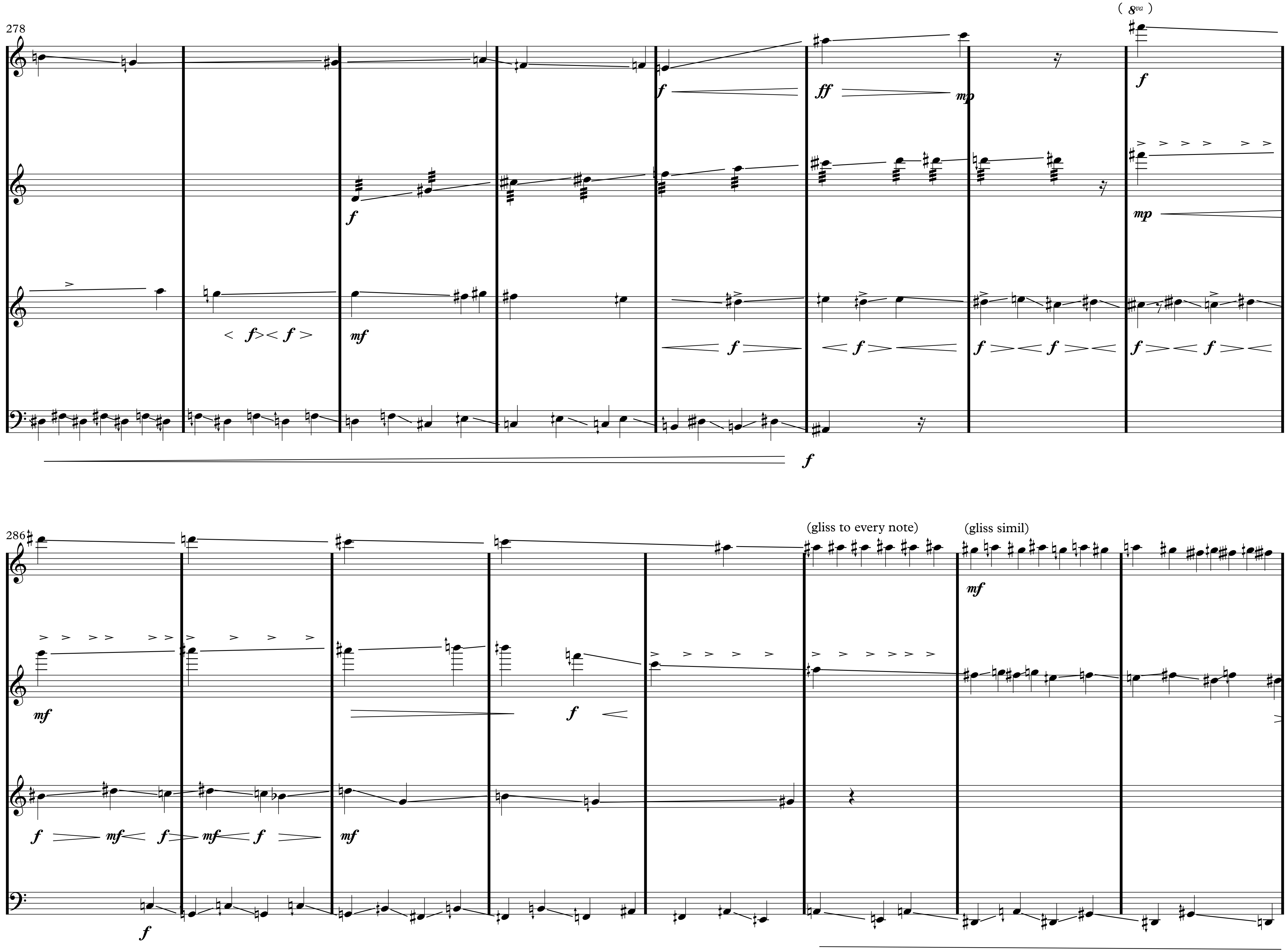

198
8va
f
mp
206
mf
mp
mf
mp
mp


214
mp
mp
mp
mf
mp
mf
mf
mp
mf
222
mf
mp
mf
f
mp
mp
mf
mp
mp
mf
mp
mf


230
mp
mf
mp
mf
mf
mp
mf
mp
mf
mf
ST
mp
ST
mp
mp
mf
238
8va
mp
mf
mp
mf
mf
mp
mf
mf
mf
mp


246
mf
mp
mf
mp
f
mf
mp
<mf><mf>
254
mf
p
f
(gliss)
(gliss)
mf
8va
f
f
p
mf


262
(8va)
8va
(gliss to every note)
(gliss simil)
(II)
270
loco
(gliss to every note)
(gliss simil)
(8va)

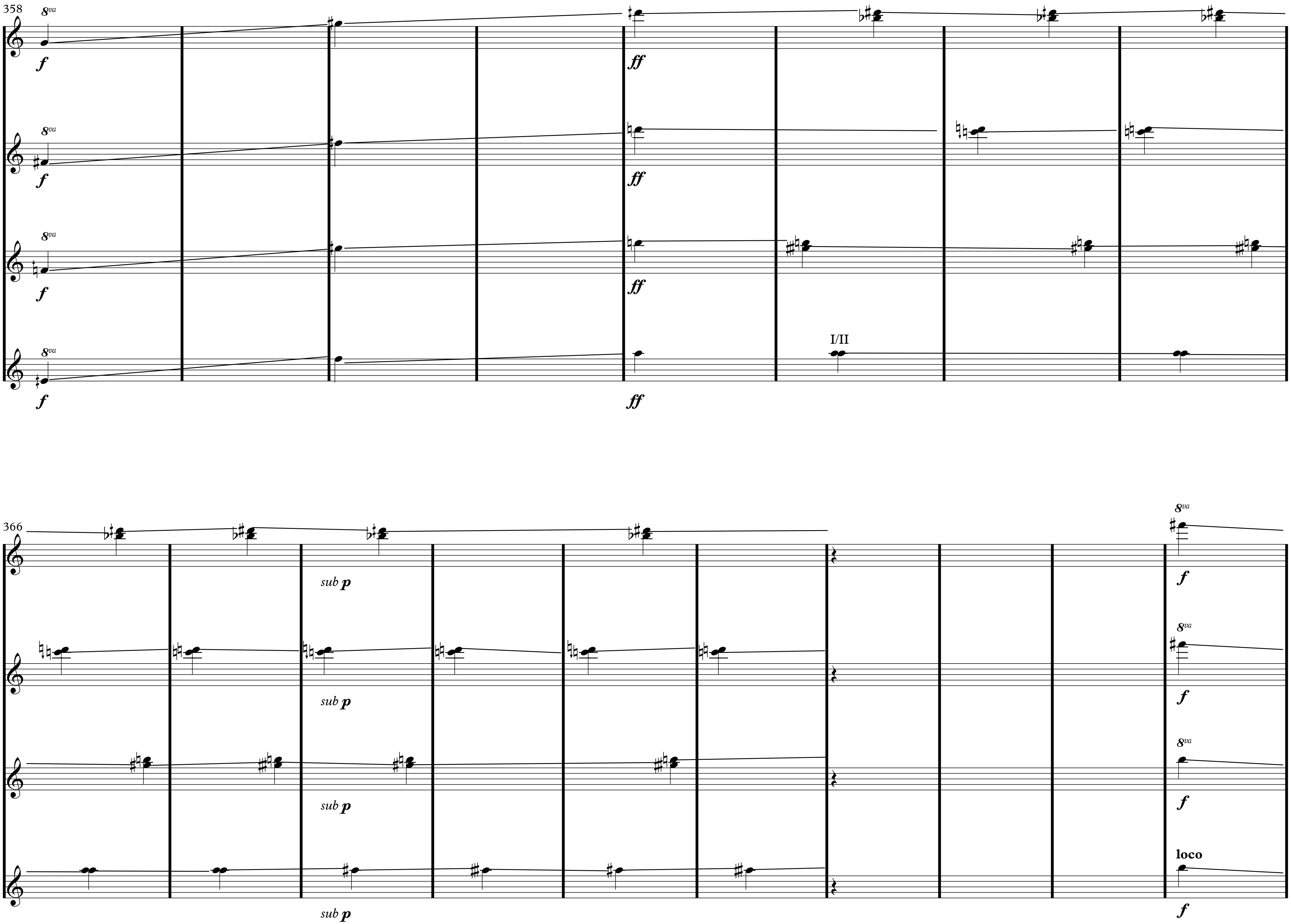

278
(8va)
286
(gliss to every note)
(gliss simil)


294
(8va)
(gliss to every note)
302
SP
loco
(gliss simil)

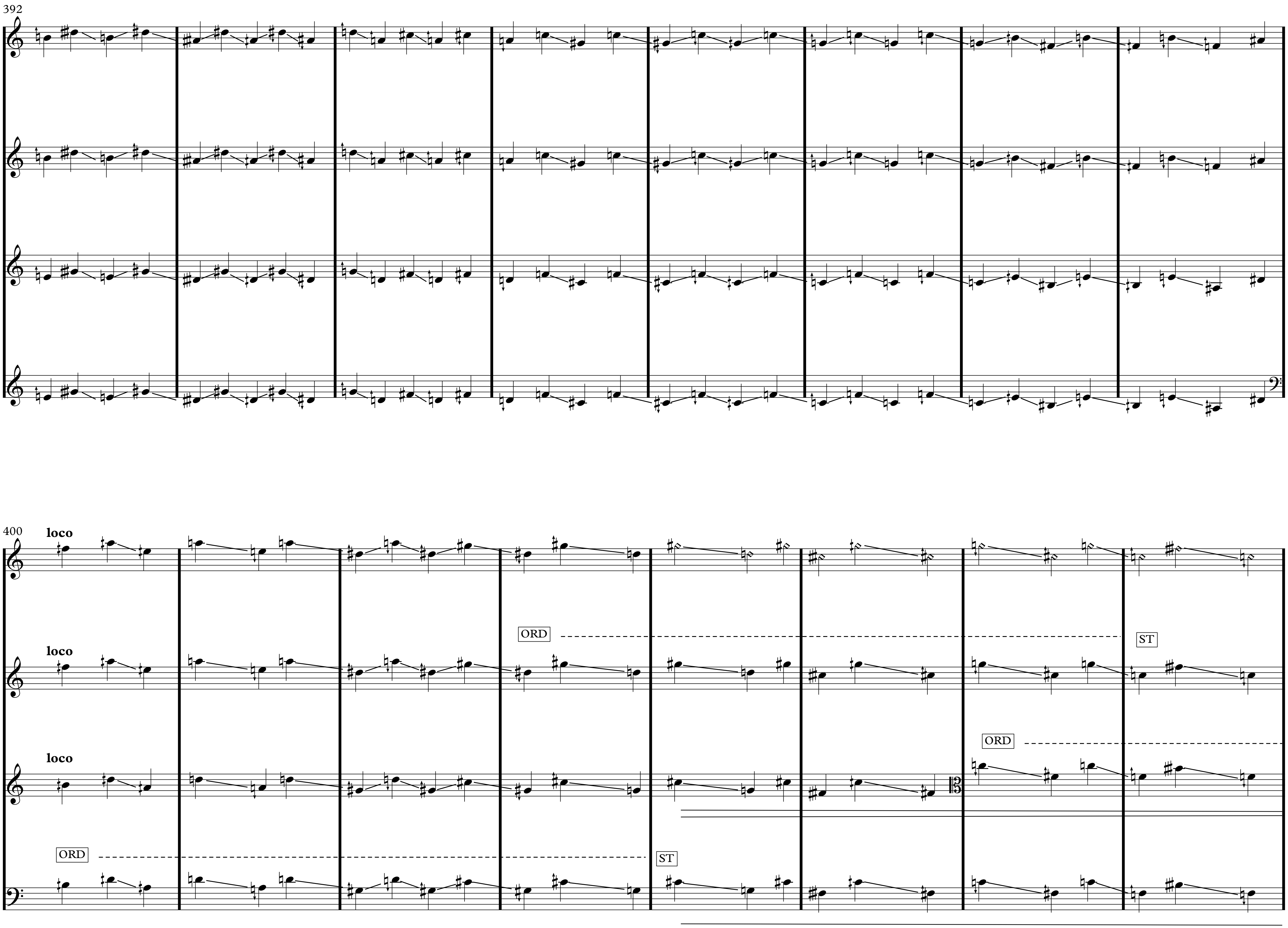

310
ORD
ff
SP
(8va)
f
ORD
SP
318
SP
(II)
ORD
8va
f
ff
SP
loco
loco
ORD


326
ORD
8va
f
SP
8va
f
ORD
8va
f
mp
334
mp
f
SP
loco
f
f
ORD
8va
f
SP

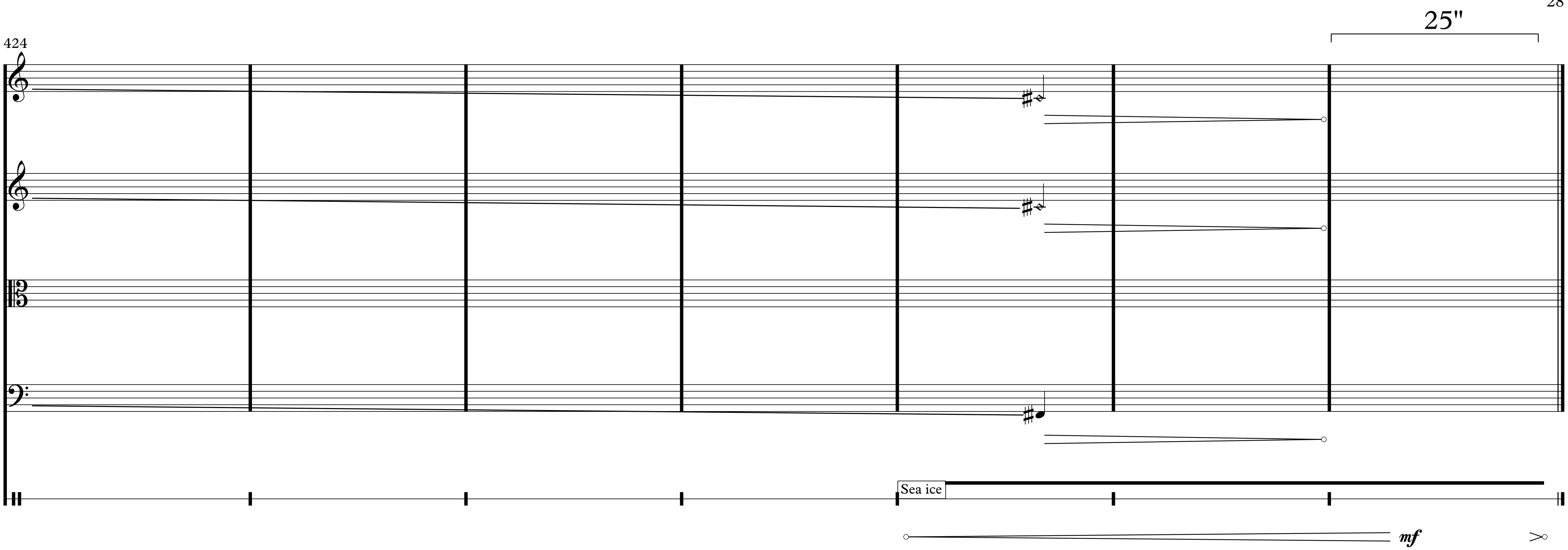

SP
loco
342
ORD
8va
f
ff
mf
ORD
8va
mp
mf
SP
f
loco
f
ORD
loco
f
8va
350
mf
f
ORD
8va
(8va)
f
ff
mf
SP
ORD
SP
ORD
f


358
8va
f
ff
I/II
366
sub p
loco


376
(gliss to every note)
(gliss simil)
mf
(gliss to every note)
(gliss simil)
mf
(gliss to every note)
(gliss simil)
mf
(gliss to every note)
(gliss simil)
mf
384


392
400
loco
loco
loco
ORD
ORD
ST
ORD
ST


408
ORD
ST
ORD
mp
mp
p
mp
p flautando
416


25"
424
Sea ice
mf