Timbral Movements in Electronic Music Composition

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Abstract

The advent of electronic music changed the way of treating timbre in compositions. The possibility of making spectrum by adding sine waves and transforming timbre by means of processing incubated the new idea “fluctuating timbre” by Gottfried Michael Koenig. On the other hand, none of the common synthesis methods are satisfactory for achieving fluctuating timbre mainly because of the lack of controllability. One of the main focuses of this research is to realize fluctuating timbre with an alternative synthesis method: band-limited oscillator. This method facilitates the systematic control of timbre by reducing the number of parameters. It is also argued that implementing fluctuating timbre and introducing relationships among multiple materials through systematic timbral movements will contribute to the musical complexity.

Keywords:
composing sound, fluctuating timbre, form, complexity, transformation, system, algorithmic composition, band-limited oscillator

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Chapter 1 Introduction

1-1 Why did I become interested in timbre?

My interest in timbre comes from my musical career as a DJ. This experience cultivated my interest in electronic music through listening and playing dance music, particularly techno, house, and trance. At first I played with turntables and later I started using a laptop for my performance. During this time, browsing online stores became a daily habit. This experience led me realize an important fact, that an overwhelming amount these tracks were inherently boring. A resounding number of them seemed to be so similar, revealing a “cut-and-paste” aesthetic, just another loop for a DJ to remix.

However on occasion I would encounter a gem, a track that would blow my mind. This led me to wonder what factors divided all of these boring tracks from the small handful of ingenious tracks. My search for answers to this question began simply by listening to many tracks as well as producing many of my own. In the process of doing this I developed my own criteria for judging dance music. This was made from criteria that was entirely subjective but included the following points:

1. Sound materials and the use of novel timbres: specifically, do the materials used in this track make me wonder how the composer made the sounds that I am hearing.
2. Continuous sonic transformation: how does the track prevent itself from having an impression of “cut-and-paste”.
3. Rich in detail: how does the track use a lot of small sounds or how does it provide interesting microtime changes.
4. Elaborate layering: how does the track give the impression that it has sufficient musical complexity.

The above criteria made me aware of my interest in timbre, sonic evolution, and complexity, which are main subjects of this thesis. Techno music has a rigid structure; rhythm is static and harmonic progressions are limited. In my opinion, the novelty of sound (timbre) and the evolution of materials over time is central to the aesthetics of this genre. Therefore, given that I possess this background, I believe it natural that my interest in timbre was initiated and nurtured through my love of dance music.

1-2 Determinants of timbre

To discuss timbre in electronic music composition, the inevitable question is “what is timbre?” The famous, or infamous definition of timbre proposed by ANSI is “that attribute of auditory sensation in terms of which a subject can judge that two sounds similarly presented and having the same loudness and pitch are dissimilar”(American National Standards Institute, 1973). Although this negative definition of timbre, which is something that is neither pitch nor loudness is far too restrictive and uninformative for the purpose of this discussion (Houtsma, 1997), it does, nevertheless, reveal a certain truth about timbre, that is that it is a musical parameter that is difficult to define. This difficulty comes from the fact that timbre is a multi-dimensional stimulus and cannot be correlated with any single physical dimension, and our subjective sound experience is that we hear this multidimensional attribute as unified perceptual objects. (Erickson, 1975).

On the other hand, the knowledge of what physical aspects affect the timbre recognition is useful for composers who work in electronic music. If we could divide this complex phenomenon called "timbre" into a number of physical factors, it enables us to some extent to deal with timbre as a set of parameters.

Tristan Jehan (2001) summarizes important factors for timbre:

1. Harmonic structure: how equally the partials are spaced.
2. The average spectral shape or how rapidly the energy dissipates from the higher partials.
3. The formant structure: the "bumpiness" of the spectrum.
4. The spectrum variations in time, especially at the attack and decay.

Points 1 to 3 of are all related to the appearance of spectrum, allowing for the determinants of timbre to be even more simplified in the following way:

1. Spectrum: harmonic structure, spectral shape, formant structure.
2. Evolution: amplitude envelope, change of spectrum in time, degree of randomness.

The idea of a "degree of randomness", did not appear in Jehan's summarization, and should therefore be added to the second criterion. Furthermore, noise should be seen as a signal that has a certain random displacement value at every moment, thus the degree of randomness is related to the amount of noise within a sound. This adheres also to the time-domain interpretation of sound that can be considered to be noise. Looking at noise, from the vantage point of the frequency domain, it can be associated with color in order to describe a particular spectral shape, this helps to articulate the way in which frequency content is statistically distributed. Therefore, nosiness can also be discussed in the context of spectrum, which is the first criterion of the above criteria.

I suggested this simple classification of the determinants of timbre because this classification is related to my discussion of "composing sound" and "complexity" which are dealt in later chapter (3-1 and 3-5).

Of course each of aforementioned factors are also impossible to be described as a single parameter. Amplitude envelope can be described with rather limited numbers of parameters such as attack, decay, sustain, release, and slope. On the other hand, spectral shape and its change over time require countless parameters to be precisely described. Therefore, extracting parameters from timbre is still not an easy task at all. Nevertheless, the above knowledge gives us a road map for the treatment of timbre in sound synthesis and transformation.

1-3 Attack and Timbre

Among the timbral factors introduced in the last section, onset or attack is examined in more detail.

David L. Wessel (1979) proposed timbre space, which is a technique to characterize timbre by placing sounds in two-dimensional space. The distance between two sounds represents the perceptual dissimilarities between two sounds. What is noteworthy is that the vertical axis is related to the centroid or mean of spectral energy distribution of the tones, and the horizontal, to the nature of the onset transient. According to Wessel, the vertical dimension is related to the perceptual brightness of the sound, and the horizontal dimension is related to the quality of the "bite" in the attack. Wessel's research suggests that, among many elements that have influence on timbral recognition, attack is one of the most dominant factors along with the distribution of spectral energy.

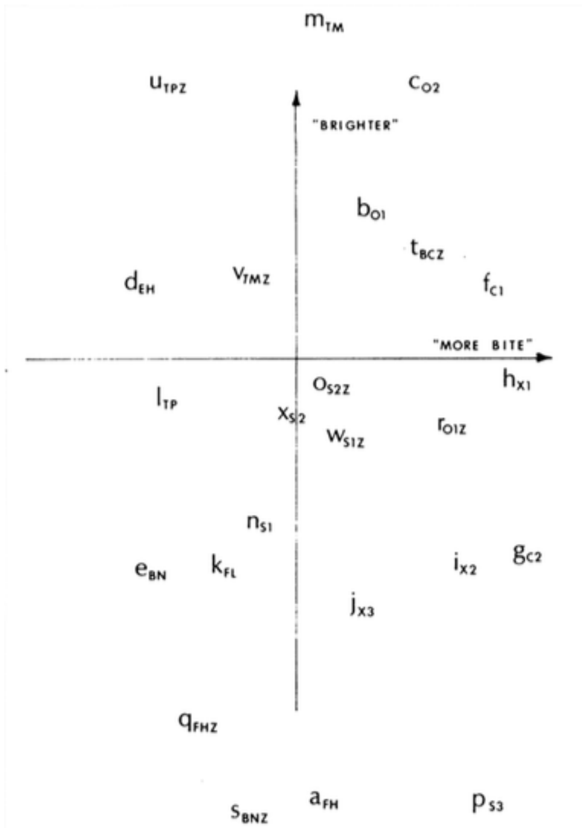


Figure 1-3: Timbre space (Wessel, 1979, p. 6)

Many other people discuss the peculiarity of attack on human perception and its importance on timbre. Trevor Wishart explains that the onset of a sound gives us some clue to the causality of the sound, such as a physical blow or a scraping contact, and we are sensitive to such onset because of the necessity for survival. The information that the onset of sound carries has a potential life-threatening importance in the species development of hearing. (Wishart, 1979)

Erickson states that our auditory apparatus is especially responsive to changes and the very beginning of the time envelope of the sound is as important as spectral envelope. Erickson explains why attack is essential to any sound being subjected to Fourier analysis. This is because, according to Fourier theory, any finite tone that has beginning and ending must contain more than a single frequency component. Even if the input is single sine tone, the onset is accompanied by other frequencies no matter how gradually a tone appears. The more sudden the onset becomes the more the frequency components spread, and we hear the broad frequency spreading as a click. (Erickson, 1975)

Undoubtedly, the pioneers of electronic music realized the importance of attack on timbre. This is evidenced in the work of Pierre Schaeffer who accidentally noticing how a bell sound, when heard after its attack, became unrecognizable as a bell. Schaeffer (as cited in Palombini, 1993) stated that this finding at the early stages of his research contained all the seeds of musique concrète. After this discovery, Schaeffer started dealing with sound recordings as compositional raw materials and focusing on their transformation (e.g. change of playback speed, backward playing, cutting, looping). (Palombini, 1993)

Koenig states, ”the *attack* is a component of the timbre.” (Koenig, 1968, p. 3) For Koenig, attack is one significant component that makes electronic music different from instrumental music. In instrumental music, “the attack characteristic for each instrument presents the greatest obstacle in the way of making timbre variable. In electronic music, by contrast, the way in which the sound is altered is important (transients).” (Koenig, 1968, p. 4)

Meanwhile, similar preoccupations about the association between attack and timbre are noticeable in some of the instrumental pieces that focus on timbre, such as the third movement of Schoenberg’s *Five Pieces for Orchestra*, Opus16, which exhibits a technique to minimize the attack characteristic of each instrument by making dynamics extremely quiet (ppp) (Erickson, 1975). This fact supports that the attack is the main obstacle for dealing with timbre in instrumental music.

Very long attack was not possible to be achieved in the instrumental music and “we have learned the musical worth of these super-long attacks.” (ibid, p. 68) Since very long attack sound is relatively new in the history of music, it seems to me that the compositional potential of such sound is not fully explored up until today.

For electronic music composition, attack is not only important for sound synthesis, but also important for sound transformation. With this in mind, in electronic music it becomes clear that careful manipulation of attack is one of the key elements for making clear transformations of timbre. If the attack characteristic is abruptly and radically changed at a certain moment of transformation, the connection between the source sound and the transformed sound can be lost. On the other hand, it is also possible to obtain a variety of sounds from a single material only by changing attack time.

1-4 Timbre in instrumental music vs. timbre in electronic music

The advent of electronic music caused a paradigm shift in the way timbre is treated in composition. The followings are some possibilities electronics brought into music:

- It became possible to make spectrum by adding sine waves.
- The amplitude envelope became controllable.
- Timbre became transformable by means of processing.

These possibilities incubated the new idea “composing sounds”. Stockhausen states,

> Every existing sound, every noise is a mixture of such sinus waves – we call it a spectrum. The numbers, intervals and dynamics of such sine waves make up the characteristics of every spectrum. They determine the timbre. For the first time the possibility was given to compose timbres in the real sense of the word, that means to compose them out of elements, thus applying the universal structural principle of a piece of music to the sound proportions as well. (Ungeheuer, 1994, p. 26)

“Composing sounds” is the concept contrasting with “composing with sounds”. In instrumental music, each instrument has particular timbre and what composers can do is to combine those predefined timbres.

> In instrumental music we do not actually indicate the "timbre" of a sound but only the "instrument" on which the sound is produced . . . At the beginning of electronic music, timbre was first concentrated on, both in the Paris studio where concrete sounds were analysed, and in Cologne where they were produced in the form of sine-wave spectra. (Koenig, 1963, p. 16)

> Instrumental music composes "with" sounds, electronic music composes sounds. (ibid, 1963, p. 2)

Regarding the different roles of timbre in the instrumental music and the electronic music, Erickson (1975) suggested another classification: “timbre as carrier” and “timbre as object” (1975). According to Erickson, the chief function of timbre has been the carrier of melodic function in most Western instrumental music: the instrument is kept the same while pitch changes (“timbre as carrier”). Many details of timbral change have been left to the performing musician as nuance. Strong contrasts in timbre, especially ones that involve changes of instrumentation, do not occur frequently. More often phrase or motive is the smallest unit of timbral change.

Musique concrète and electronic sound generation has led composers toward music that depends more on contrasts of timbre contrasts than on constancy (“timbre as object”). The pitch/melodic part of the music became less important.

> Gross changes of pitch register, dynamics or articulation, and the avoidance of stepwise motion and rhythmic regularity enhance the perception of a sequence of contrasting sound objects. (Ericsson, 1975, p.13)

Certainly there were attempts to explore timbral contrasts before the birth of electronic music, as is evidenced by the third movement of Schoenberg's *Five Pieces for Orchestra*, Opus16. However, the fast rate of timbral change became commonplace only after the technical means for creating controlled timbral sequencing became more readily possible to realize.

"Timbre as carrier" can be paraphrased as a compositional method to suppress the timbre movements and move other musical parameters (e.g. pitch, rhythm). "Timbre as objects" can be paraphrased as a compositional method to move the timbre and suppress the other musical dimensions. The detailed techniques to reduce pitch movements and rhythm changes while moving timbre will be examined in chapter 3-6.

1-5 Fluctuating Timbre

Another important concept Koenig suggested regarding timbre in electronic music is "bewegte Farbe" (fluctuating timbre). (Koenig, 1992)

In the end of 1940s, the German physicist Meyer-Eppler conducted acoustic research and demonstrated various sound transformations. For example, he showed the transition from sine wave to noise using amplitude modulation. This demonstrated how filtering white noise can the reversal of this process, in addition he also showed how the transformation from a sequence of impulses into a sound with pitch could be realized by continual acceleration. This research influenced composers of those days, and it became the foundation of some electronic pieces such as Stockhausen's *Gesang der Junglinge*. The possibilities of sound transformation let Koenig conceive the concept of fluctuating timbre. (Ungeheuer, 1994) Koenig states, "One hoped for (and categorically demanded) an unbroken continuum of all timbres; not only of all timbres, but the continuum between the timbre, stationary in itself, and the musical structure. The aim was the contoured, the fluctuating timbre." (Ungeheuer, 1994, p.30)

Koenig also made the following classification of electronic sounds. (Koenig, 1963)

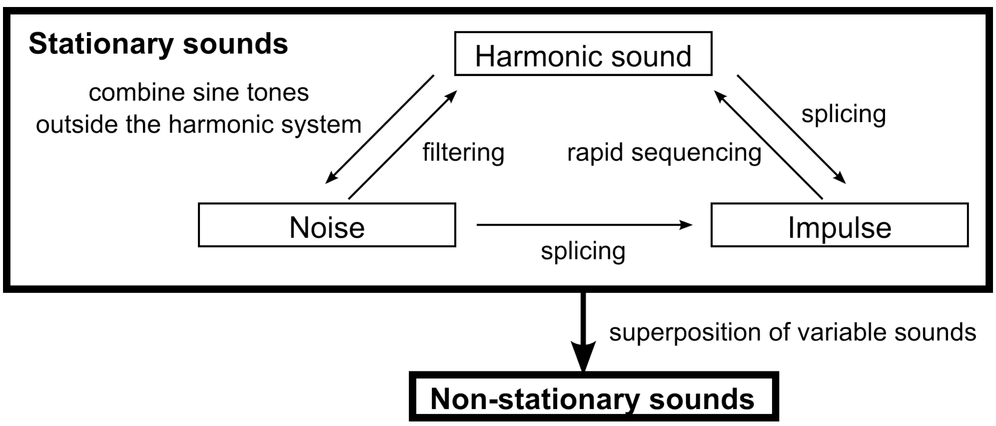


Figure 1-5: Illustrating Koenig's classification of electronic sound

Electronic sounds can be divided into two families: stationary sounds and non-stationary sounds. Stationary sounds consist of harmonic sound, noise, and impulse. As Mayer-Eppler demonstrated, one of those sounds can be transformed into a different type of sound. Noise-like sounds can be obtained from sine tones not only by means of amplitude modulation, but also by adding sine tones outside of the harmonic structure. Impulse-like sounds can be obtained from continuous sounds by means of splicing. According to Koenig, the stationary sound is complete in itself and timbre stays constant. On the other hand, non-stationary sounds have no constant timbre within a specified duration. These types of sounds therefore, according to Koening, often occur as a result of superposition of continuous but variable sounds. (Koenig, 1963)

Of course, there were a lot of technical limitations to achieve continuous timbre movements in those days. Nowadays, the development of digital signal processing allows us to achieve a variety of continuous timbre movements that were unavailable in 1960s. Thus, I believe it is worthwhile to re-examine the concept of fluctuating timbre today.

1-6 Sound transformation and form

In reference to the discussion of "composing sound" and "fluctuating timbre", there is another subject that needs to be considered: "form".

In Cologne studio, the creation of form is closely related to the act of making materials and processing materials. Stockhausen considered that "composing sound was directly linked with composing form; a sound could only perform its function in the place it was conceived for." (Koenig, 1987, p. 165) Therefore, Stockhausen did not reuse the sound he made in any other place in any other piece. According to Koenig, form was seen as an automatic consequence of the treatment of material. "Form could thus only be discussed as the properties of the material into which it had breathed life". (ibid, p. 172)

Koenig's also states, "the individual sound was regarded as something formed". (ibid, p. 166) This statement connects the discussion of "composing sound", "fluctuating timbre", and "form". According to Koenig, the actual tone and the figurative form are distinctly separated in instrumental music (Koenig, 1992). Form can be only emerged as a result of the combination of more than one note. In electronic music, on the other hand, a single sound can be a form. The reason is that timbre movement can be implemented inside this single sound. In other words, timbre movement articulates the form in the electronic music.

Koenig remarks, "Form as I see it . . . is, rather, the way in which music is experienced in time". (Koenig, 1987, p. 171) If a sound is non-stationary, we can experience it with the passage of time. Thus, making timbral fluctuation in electronic music able to be interpreted as the act of making time; accordingly it can also be interpreted as the act of making form.

The above discussion concerns the creation of micro-structure. In other words, this discussion is based upon the question regarding how instrumental experience, in macro-time (rhythmic relationships among parameter values), can be transferred to micro-time (timbre formation laws). (Koenig, 1963) Meanwhile, Koenig's idea about the formation of macro structure is also worth examining.

> My electronic compositions have always been prompted by the technical facilities in a studio. I was out to use the machines not just in a rational but in a musical fashion. they were meant to assume formative tasks, the way musicians do. By 'formative' I mean the creation of neighbouring relationships, vertical and horizontal. (Koenig, 1987, p. 168)

In this sentence, Koenig apparently concerns the macro form. The important point is that Koenig placed a high premium on making relationships between sound structures to achieve the unity. Vertical relationships occur among simultaneously played structures. Horizontal relationships occur in the longer time span.

> Neighbouring relationships, which give a work its aesthetic unity, can be penetrated by individual characters sounding like comments on the basic thesis, but also enhancing the common aspect of the material responsible for the unity . . . This aesthetic task of obtaining variety from unity resulted in workable techniques of sound production and sound derivation. (ibid)

Koenig made use of the sound transformation as a way of generating multiple variants that have mutual relationships. For example, Koenig repeated generations of transformation of basic sound in *Terminus,* and made a tree-like structure (the generation diagram Koenig used in this piece is shown in chapter 3-4-3).

> The compositional principle here consists solely of the systematic derivation of material structures which due to the mechanical procedure are structurally related; neighbouring relationships are not formally established but appear as the derived materials are presented — successively or simultaneously. (Koenig, 1987, p. 170)

This organizational approach of composition is different from the normal time line composition. This scheme enabled the composer not only to increase the sonic variety, but also to create the unity by clarifying the relationships among materials.

> It was to be expected that the sound material would first differ because of transformation, but would later forfeit these differences because common characteristics caused by transformations dominate the original characteristics of the "Urklänge". （Koenig, 1971, p.8）

This strategy of deriving many variants that are supposed to enhance the unity of the piece is inherited to Koenig's program for instrumental composition "project 1". (Koenig, 1987)

Making variants that have neighbouring relationships enhances not only the unity but also the complexity. This subject will be discussed in detail in chapter 3-5.

1-7 Potential and limitation of electronic sound

Whereas musique concrète in Paris and electronic music in Cologne opened up a new horizon for composition, there was dissatisfaction with the musical quality they could achieve among some composers. Jean-Claude Risset explains why he was disappointed with them:

> To put things simply, I found that although musique concrète was opening the scope of musical sounds, it did not provide to the composer with the refined compositional control he could exert when writing instrumental music. Musique concrète offers a rich variety of powerful sounds: however the possibilities of transforming those sounds are rudimentary in comparison, so that it is hard to avoid an aesthetics of collage. In contradistinction, one could control the sounds of electronic music more precisely, but these sounds tend to be so simple and dull that one is tempted to enrich them by complex manipulations, thus destroying the effectiveness of control one could have over them. (Risset, 2003, p. 1)

Risset's dissatisfaction with musique concrète and early electronic music prompted him to explore the computer sound synthesis. Risset made an effort to synthesize the brassy sound that has the same richness as the real instrument sound. Every partial of the real instrument sound has different attack and decay time. In other words, the spectrum of acoustic instruments exhibits micro time changes. Because of this fact, Risset started applying a different envelope function to each partial.

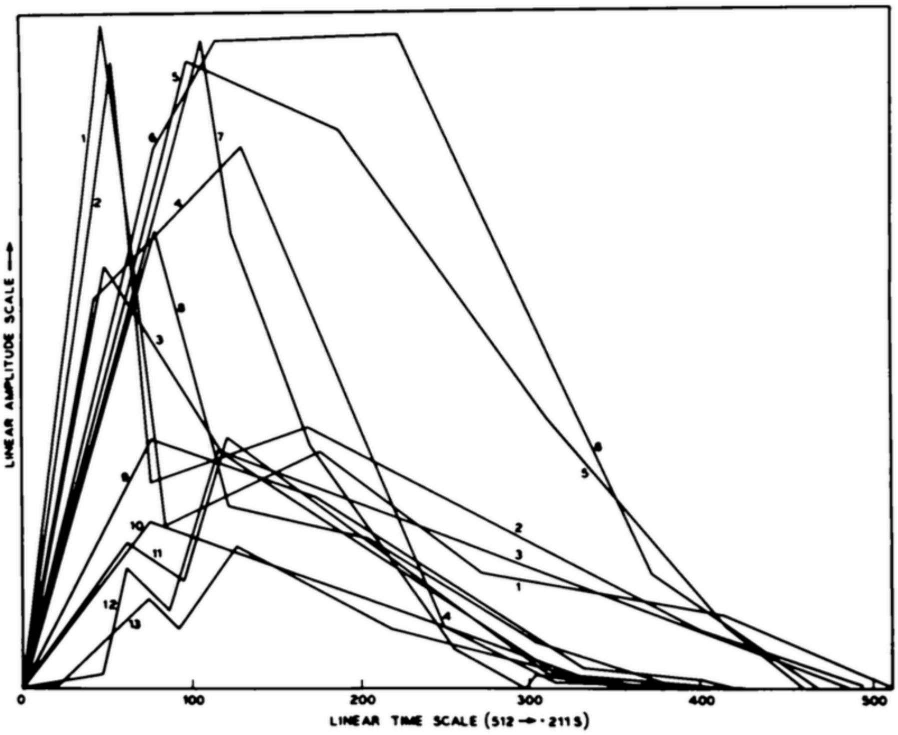


Figure 1-7: Line-segment functions that approximate the evolution in time of 13 harmonics of a D4 trumpet tone lasting 0.2 sec. (Risset, 1985, p. 12)

In addition, Risset found through the analysis of trumpet sound that the essential characteristic of brass tone was the fact that the spectrum varies with loudness: when the loudness increases, the proportion of high frequency energy also increases. (Risset, 1985)

The problems Risset pointed out remain until today despite of the technical development. Concrete sounds and basic oscillator sounds are still the main materials for electronic music compositions. Although the drawbacks of concrete sounds and oscillator sounds that Risset argued might not be a significant issue for some composers, I personally agree with Risset's frustration, especially since I have experienced the difficulty of logically constructing sounds — by way of using concrete materials — due to their inherent lack of controllability. Additionally, I have also frequently experienced the situation, as a listener, that the static qualities of oscillator sounds make music dull. Changing the amplitude envelope of each partial is surely a reasonable approach to making sounds more rich and sonorous, but how should we control the vast number of envelopes? Relating amplitude and spectral shape seems to make convincing sounding results, but how could we define this relationship efficiently? I am trying to answer these questions to certain extent through the development of my own synthesis program that will be explained in the next chapter.

Chapter 2 Synthesis for fluctuating timbre

2-1 Problems of existing synthesis techniques

According to Risset, standard oscillator sounds are static and result in sounds that have an overall flatness or dullness. On the other hand, newer synthesis techniques produce lively timbral evolution, but also have their own problems. In this chapter, existing synthesis methods and their problems are first overviewed. After this, I will focus on a band-limited oscillator that I consider to be an efficient new synthesis technique for making fluctuating timbre. Regarding this band-limited oscillator, the basic principle, specifically being the implementation of timbral movements and their overall compositional applications, will be discussed.

In 1969, Max Mathews pointed out the two fundamental problems in sound synthesis:
"(1) the vast amount of data needed to specify a pressure function--hence the necessity of a very fast program--and (2) the need for a simple, powerful language in which to describe a complex sequence of sounds." (as cited in Smith, 1991)

Julius O. Smith considers that problem (1) has been solved to a large extent by the development of digital processor performance, however, problem (2) cannot be completely solved. Smith states that it takes millions of samples to make a sound for a musical piece and thus sound samples must be synthesized algorithmically, or derived from recordings of natural phenomena. (ibid)

In light of this position, Smith proposed the following taxonomy of digital synthesis techniques.

Processed Recording	Spectral Model	Physical Model	Abstract Algorithm
Concrète	Wavetable F	Ruiz Strings	VCO,VCA,VCF
Wavetable T	Additive	Karplus-Strong Ext.	Some Music V
Sampling	Phase Vocoder	Waveguide	Original FM
Vector	PARSHL	Modal	Feedback FM
Granular	Sines+Noise (Serra)	Cordis-Anima	Waveshaping
Prin. Comp. T	Prin. Comp. F	Mosaic	Phase Distortion
Wavelet T	Chant		Karplus-Strong
	VOSIM		
	Risset FM Brass		
	Chowning FM Voice		
	Subtractive		
	LPC		
	Inverse FFT		
	Xenakis Line Clusters		

Figure 2-1: A taxonomy of digital synthesis techniques by Smith (1991)

Some of the techniques Smith mentioned are widely used and some others are not. Although I have used only a part of Smith's list of techniques, I see that most synthesis has either or both of the following two problems, with respect to how changes of timbre can be constructed. First, they can only produce limited qualities of sounds. Second, it is difficulty to control the change of timbre in many cases.

The problems of some popular synthesis methods regarding the generation of timbre movements are concretely explained below.

Changing the cutoff frequency of filter in subtractive synthesis (VCO, VCA, VCF) seems to be the most easily and malleable way of changing timbre, but the sonic variations this method can make are limited. Of course, it is possible to combine a multiple of oscillators and filters, but as a consequence controllability will be lost.

FM synthesis became one of the most popular synthesis methods probably because of its potential for generating a great variety of different sounds with limited number of parameters. Nonetheless, FM synthesis cannot be considered as intuitive enough in terms of controlling timbre because the amplitudes of the carrier and sideband components are determined by Bessel functions. (Chowning, 1973) Although organizing FM parameters into harmonicity ratio and modulation index makes them somewhat more manageable to deal with such timbral movements, the change of parameter values as well as the resultant alteration of sound, still does not result in a linear relationship.

This is also seen to a similar extent with granular synthesis, which is a useful technique for generating lush sounds with microtime timbral fluctuation. Granular synthesis also makes it possible to dynamically modify sound by changing parameters and reading various positions of a sample (in other words, a waveform of each grain). However, it has also limited controllability because its resultant sound cannot be described as a set of equations and is thus rather unpredictable.

FM and granular synthesis also have particular qualities in sound and timbral evolution. I have heard claims such as "I don't like FM sounds" or "sounds made with granular synthesis are always similar". I personally think that the unique quality of a specific synthesis procedure is not a problem because such a quality can be useful in certain compositional contexts. However, depending on one's viewpoint, this fact can be considered as an overall limitation of these types of synthesis techniques.

Additive synthesis is another type of synthesis that can theoretically produce any sound and yield a very detailed timbral evolution — achieved by specifying different amplitude envelopes for different partials. This is the method in which Risset applied to synthesize instrument-like sounds. However, it is an enormous task for a composer to determine all the envelope specifications for generating even one single sound.

Due to this aforementioned problems of controllability in existing synthesis methods, intuitive and predictable manipulations of timbre have not been easily achieved. I have speculated that this is one reason why the systematic treatment of timbre has not been fully explored, not only in the instrumental music, but also in electronic music.

In the next section, I would like to suggest the use of band-limited oscillators as an alternative synthesis method, one that allows for both the intuitive control of timbre and facilitating variations of resultant sounds. I also regard physical modeling as another useful synthesis technique that can produce rich timbral evolution with limited amount of parameters. Although physical modeling is not covered in this thesis, I am planning to explore its potential in future.

2-2 Band-limited oscillator: overview

Band-limited oscillator, as its name suggests, is an oscillator that produces waveforms whose partials appear only within a limited frequency band. Already in 1976, Moorer describes the principle of what is then called “discrete summation” as a means to synthesize complex audio spectra with only a limited set of phase-coupled oscillators, and the subject is also mentioned in the book of Tempelaars. (Tempelaars, 1996, p.291) The underlying principle of geometric series summation is already treated by Euclid in one of his *Elements*. Peter Pabon demonstrated the principle of the band-limited oscillator in his Analysis-resynthesis class at the Institute of Sonology and had a renewed interest in using this old scheme. In his research on singing voice quality, Pabon discovered that in the voice spectrum the level contrasts were actually much steeper than the general filter model predicted, and that these contrasts were due to steep cut-off filter slopes that are not linked to sharp filter resonances. To give a sound this more "natural" touch an additive scheme with phase coupled band-limited oscillators is to be preferred above a subtractive filter scheme as with such a scheme steep cutoff slopes are generally associated with sharp resonances. (Pabon, personal communication, May 4, 2015)

In addition to the fact that band-limited oscillator is suitable for voice synthesis, I found it very useful for electronic music composition in general. Therefore, I made the implementation of band-limited oscillator UGen for a SuperCollider environment. I thereby extended the synthesis scheme with some extra controls that allow a fluent transition to a band-limited odd-harmonic spectrum, and the option to gradual fade in or out an extra harmonic, options that were missing in the original concept. From here, “Band-Limited Oscillator” with capitalized initials indicates the UGen I made, while “band-limited oscillator” with lowercase letters indicates the general synthesis technique.

BLOsc

BLOsc is the first version of my implementation of Band-Limited Oscillator. BLOsc can be played with either audio rate (BLOsc.ar) or control rate (BLOsc.kr). There are following arguments in this UGen.

freq	Fundamental frequency in Hertz.
loHarmonics	The lowest harmonic index.
numHarmonics	The total number of harmonics.
slope	The slope of spectrum. It should be > 0.
	< 1: The lower the harmonics, the higher the amplitude.
	= 1: Flat spectrum.
	> 1: The higher the harmonics, the higher the amplitude.

evenOddRatio	The amplitude ratio of harmonics of even indices to those of odd indices. It should fall between 0 and 1. The lower the value, the more square-wave like, or triangle-wave like quality of sound comes out.
mul	Output will be multiplied by this value.
add	Output will be added to this value.

freq, *slope*, and *evenOddRatio* can be modulated with either audio rate or control rate UGens. While arguments *loHarmonics*, *numHarmonics*, and *slope* came from Pabon's idea, implementing *evenOddRatio* is my own ideas. This means *evenOddRatio* is meant to be designed for compositional purpose rather than voice synthesis purpose. The default setting of arguments is:
freq: 440, *loHarmonics*: 1, *numHarmonics*: 15, *slope*: 1, *evenOddRatio*: 1, *mul*: 1, *add*: 0.

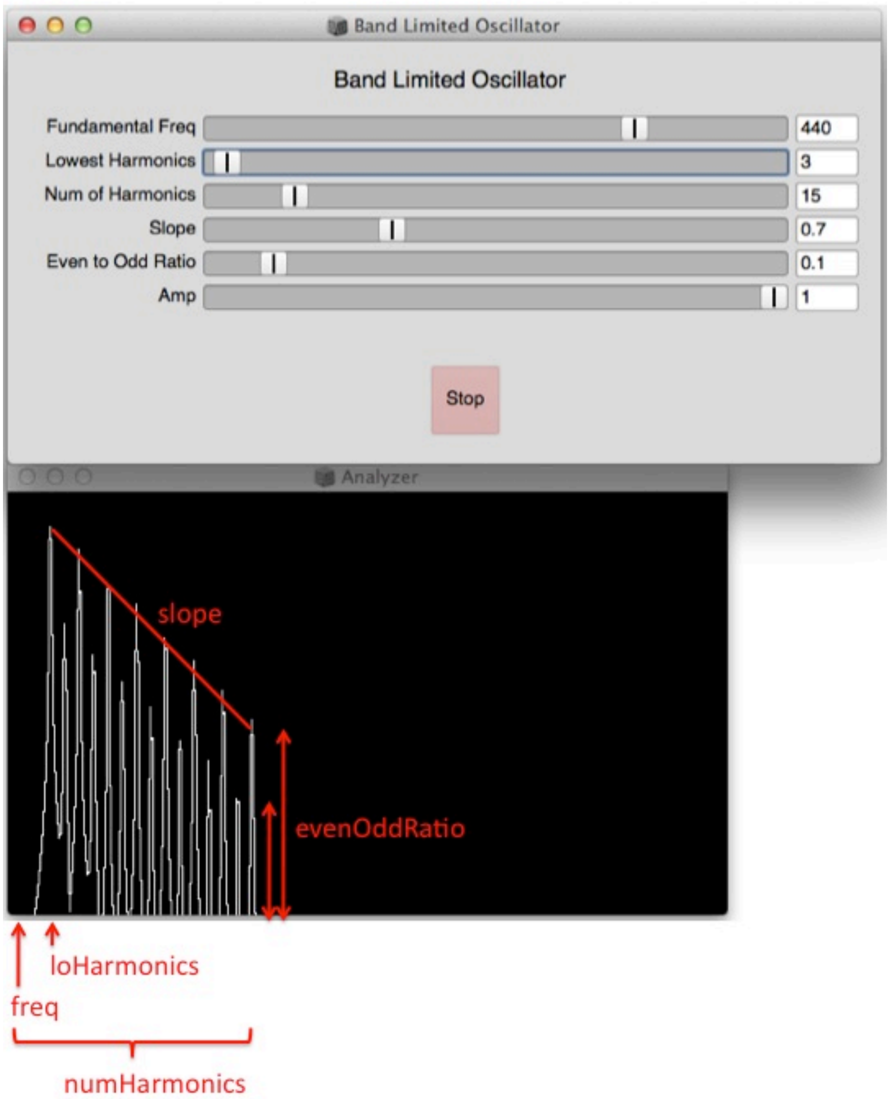


Figure 2-2: Controlling the parameters of BLOsc.ar with GUI (top) and the spectrum achieved by this specific parameter setting (bottom)

BLOsc2

BLOsc2 is the improved version of Band-Limited Oscillator. Different from BLOsc, *hiHarmonics* is given as an argument instead of *numHarmonics*, and *loHarmonics* and *hiHarmonics* accept floating-point values as inputs. For example, if *loHarmonics* is 1.3, it makes an interpolated state between *loHarmonics* = 1 and *loHarmonics* = 2. Floating point input for *hiHarmonics* works in the same manner. This change makes it possible to modulate *loHarmonics* and *hiHarmonics* continuously. On the other hand, the CPU usage of BLOsc2 is heavier than that of BLOsc. If modulations of *loHarmonics* and *hiHarmonics* are not necessary, the use of BLOsc is recommended instead of BLOsc2. Arguments of BLOsc2 are as follows.

freq	Same as BLOsc.
loHarmonics	The lowest harmonic index (starting from 1.0). Floating-point value can be given and is modulatable.
hiHarmonics	The highest harmonic index. Floating-point value can be given and is modulatable.
slope	Same as BLOsc.
evenOddRatio	Same as BLOsc.
mul	Same as BLOsc.
add	Same as BLOsc.

Advantages of Band-Limited Oscillator in composition

The followings are some advantages of Band-limited oscillator in terms of making variable timbre for compositional purpose.

- The scheme of band-limited oscillator can be considered as a special case of additive synthesis. In my Band-limited Oscillator implementation, one parameter (e.g. *slope*, *evenOddRatio*) controls amplitude values of multiple partials at the same time, thus the number of parameters is dramatically reduced compared with the conventional additive synthesis models requiring specification for every single partial. Whereas it became impossible to draw elaborate spectral shapes or specify unique amplitude envelope for each partial, as a trade-off, the controllability became extremely enhanced.

- The change of parameter values and the change of resultant timbre have perceivable and nearly linear relationship, thus enabling composers to intuitively control timbre.

- Modulating parameters with other signals can dynamically change the spectrum, thus the fluctuating timbre can be easily implemented. Simultaneous modulations of multiple parameters, either with high frequency control signals or with low frequency control signals, make it possible to generate a variety of interesting sound movements that cannot be obtained with other synthesis methods.

2-3 Band-limited oscillator: principle

In this section, the mathematical principle of band-limited oscillator is explained.

Output values of generated signals are calculated using the formula for the summation of geometric series. For $r \neq 1$, the formula for the summation of the first n terms of a geometric series is:

$$
\begin{aligned}
1 + r + r^2 + r^3 + \cdots + r^{n-1} &= r^0 + r^1 + r^2 + r^3 + \cdots + r^{n-1} \\
&= \sum_{k=0}^{n-1} r^k = \frac{1 - r^n}{1 - r}
\end{aligned}
$$

This formula can be derived as follows:

$$s = 1 + r + r^2 + r^3 + \cdots + r^{n-1}$$

$$sr = r + r^2 + r^3 + r^4 + \cdots + r^n$$

$$s - sr = 1 - r^n$$

$$s(1 - r) = 1 - r^n$$

$$s = \frac{1 - r^n}{1 - r} (\text{if } r \neq 1)$$

For $r \neq 1$, the generalized summation formula for a geometric series is:

$$
\begin{aligned}
r^a + r^{a+1} + r^{a+2} + r^{a+3} + \cdots + r^b &= \sum_{k=a}^{b} r^k \\
&= \frac{r^a - r^{b+1}}{1 - r}
\end{aligned}
$$

where $a, b \in \mathrm{N}$

This formula can be derived as follows:

$$
\begin{aligned}
\sum_{k=a}^{b} r^k &= \sum_{k=0}^{b} r^k - \sum_{k=0}^{a-1} r^k \\
&= \frac{1 - r^{b+1}}{1 - r} - \frac{1 - r^a}{1 - r} \\
&= \frac{1 - r^{b+1} - 1 + r^a}{1 - r} \\
&= \frac{r^a - r^{b+1}}{1 - r}
\end{aligned}
$$

For $r \neq 1$, the sum of a geometric series containing only even powers of r is:

$$
\begin{aligned}
1 + r^2 + r^4 + r^6 + \cdots + r^{2n} &= r^0 + r^2 + r^4 + r^6 + \cdots + r^{2n} \\
&= \sum_{k=0}^{n} r^{2k} = \frac{1 - r^{2n+2}}{1 - r^2}
\end{aligned}
$$

This formula can be derived as follows:

$s = 1 + r^2 + r^4 + r^6 + \cdots + r^{2n}$

$sr^2 = r^2 + r^4 + r^6 + r^8 + \cdots + r^{2n+2}$

$s - sr^2 = 1 - r^{2n+2}$

$s(1 - r^2) = 1 - r^{2n+2}$

$s = \dfrac{1 - r^{2n+2}}{1 - r^2}$ (if $r \neq 1$)

For $r \neq 1$, the generalized summation formula for a geometric series containing only even powers of r is:

$$\begin{aligned} r^{2a} + r^{2(a+1)} + r^{2(a+2)} + r^{2(a+3)} + \cdots + r^{2b} &= \sum_{k=a}^{b} r^{2k} \\ &= \frac{r^{2a} - r^{2b+2}}{1 - r^2} \end{aligned}$$

where $a, b \in \mathrm{N}$

This formula can be derived as follows:

$$\begin{aligned} \sum_{k=a}^{b} r^{2k} &= \sum_{k=0}^{b} r^{2k} - \sum_{k=0}^{a-1} r^{2k} \\ &= \frac{1 - r^{2b+2}}{1 - r^2} - \frac{1 - r^{2(a-1)+2}}{1 - r^2} \\ &= \frac{1 - r^{2b+2} - 1 + r^{2a}}{1 - r^2} \\ &= \frac{r^{2a} - r^{2b+2}}{1 - r^2} \end{aligned}$$

Now, suppose we would like to calculate the signal function for the following spectrum. (Figure 2-3-1)

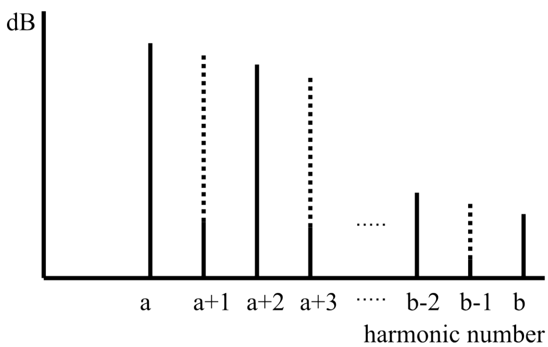


Figure 2-3-1: Spectrum to be obtained

Before dealing with the above signal, I would like to discuss simpler spectrum that all the harmonics have equal amplitude. (Figure 2-3-2)

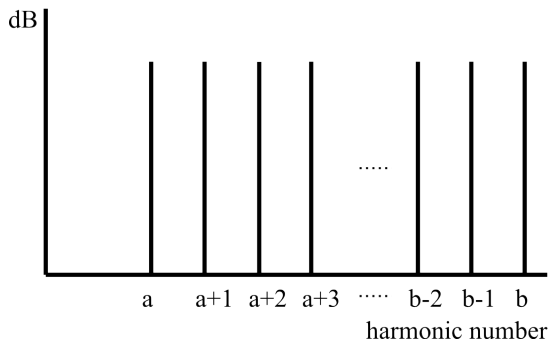


Figure 2-3-2: Flat band-limited spectrum

Here, substitute r of the following expression with $r = e^{i\varphi}$.

$$\begin{aligned} r^a + r^{a+1} + r^{a+2} + r^{a+3} + \cdots + r^b &= \sum_{k=a}^{b} r^k \\ &= \frac{r^a - r^{b+1}}{1-r} \\ &= \frac{e^{ia\varphi} - e^{i(b+1)\varphi}}{1-e^{i\varphi}} \end{aligned}$$

where $a,b \in \mathrm{N}$

$e^{i\varphi}$ represents the sine oscillator whose instantaneous phase equals to φ.

Next, the following spectrum that dB goes down linearly when harmonic number goes higher is examined. (Figure 2-3-3)

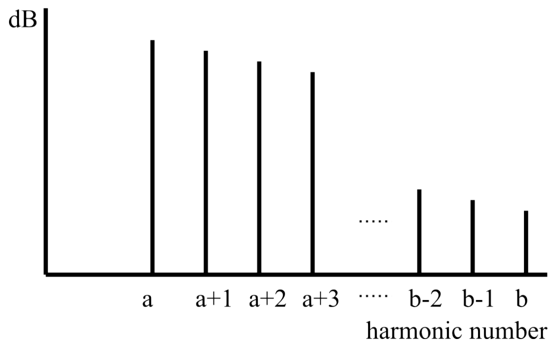


Figure 2-3-3: Sloped band-limited spectrum

This time, substitute r of the following expression with $r = pe^{i\varphi}$

$$
\begin{aligned}
r^a + r^{a+1} + r^{a+2} + r^{a+3} + \cdots + r^b &= \sum_{k=a}^{b} r^k \\
&= \frac{r^a - r^{b+1}}{1 - r} \\
&= \frac{p^a e^{ia\varphi} - p^{(b+1)} e^{i(b+1)\varphi}}{1 - pe^{i\varphi}}
\end{aligned}
$$

where $a, b \in \mathrm{N}$

The amplitude of the harmonic number $a, a+1, a+2, a+3 \cdots$ are $p^a, p^{a+1}, p^{a+2}, p^{a+3} \cdots$. Making the vertical axis the logarithmic scale, this geometric series of amplitude value is described as a linear slope because:

$$\log p^a = a \log p$$

$$\log p^{a+1} = (a+1) \log p$$

$$\log p^{a+2} = (a+2) \log p$$

$$\log p^{a+3} = (a+3) \log p$$

…

The dB scale is actually a log scale.
The formula of dB:

$$dB = 20 \cdot \log_{10}\left(\frac{A_1}{A_{ref}}\right)$$

A_1 : measured amplitude

$\mathrm{A}_{\mathrm{ref}}$: reference amplitude

Therefore, the amplitude series of $p^a, p^{a+1}, p^{a+2}, p^{a+3} \cdots$ is depicted as a linear slope in dB scale. p is the slope of the amplitude in dB scale.

Moving on from here to the calculation of band-limited spectrum means that the amplitude ratio of even harmonics to odd harmonics can be manipulated. Depending on whether the lowest and the highest harmonic indices are even or odd number, there are four possible cases.

(1)

Figure 2-3-4: Lowest harmonic: odd, Highest harmonic: odd

(2)

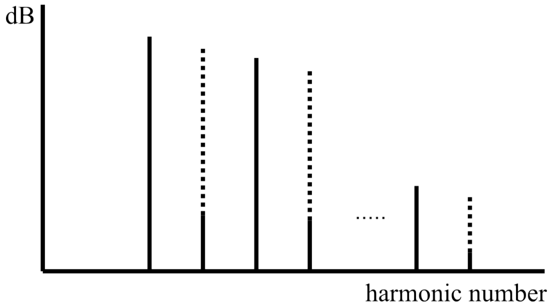


Figure 2-3-5: Lowest harmonic: odd, Highest harmonic: even

(3)

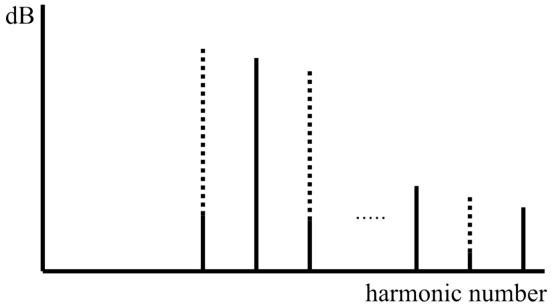


Figure 2-3-6: Lowest harmonic: even, Highest harmonic: odd

(4)

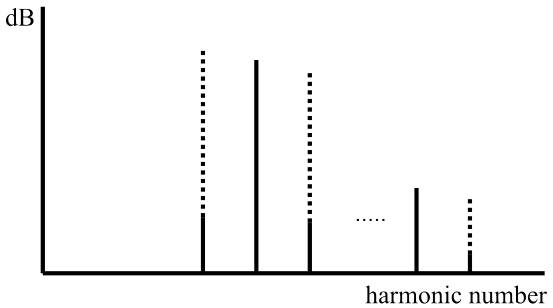


Figure 2-3-7: Lowest harmonic: even, Highest harmonic: even

The following symbols are used for the subsequent expressions:
φ = instantaneous phase
Sl = slope (argument)
Lo = the lowest harmonic index (argument)
Hi = the highest harmonic index
Loe = the lowest even harmonic index
Hie = the highest even harmonic index
Ef = evenOddFactor = 1 – evenOddRatio (argument)

The total number of harmonics is provided as an argument (*numHarmonics*) instead of the highest harmonic index. The highest harmonic index is calculated with the following expression.
$Hi = Lo + numHarmonics - 1$
The calculations of the lowest even harmonic index and the highest even harmonic index are as follows.
If *Lo* is even, *Loe* = *Lo*. If *Lo* is odd, *Loe* = *Lo* + *1*.
If *Hi* is even, *Hie* = *Hi*. If *Hi* is odd, *Hie* = *Hi* - *1*.

The output values of the signal can be obtained as complex numbers with the expression below. Using Euler's formula, the complex number calculation can be written as sine and cosine calculation.
Euler's formula:
$e^{i\varphi} = \cos\varphi + i\sin\varphi$

output signal as a complex number = $\dfrac{r^{Lo} - r^{Hi+1}}{1-r} - Ef\,\dfrac{r^{Loe} - r^{Hie+2}}{1-r^2}$

$r = Sl\cdot e^{i\varphi}$

$$\frac{Sl^{Lo}\cdot e^{iLo\varphi} - Sl^{Hi+1}\cdot e^{i(Hi+1)\varphi}}{1 - Sl\cdot e^{i\varphi}} - Ef\,\frac{Sl^{Loe}\cdot e^{iLoe\varphi} - Sl^{Hie+2}\cdot e^{i(Hie+2)\varphi}}{1 - Sl^2\cdot e^{i2\varphi}}$$

$$= \frac{Sl^{Lo}\cdot\left[\cos(Lo\cdot\varphi) + i\sin(Lo\cdot\varphi)\right] - Sl^{Hi+1}\cdot\left\{\cos\left[(Hi+1)\varphi\right] + i\sin\left[(Hi+1)\varphi\right]\right\}}{1 - Sl\cdot(\cos\varphi + i\sin\varphi)}$$

$$- Ef\,\frac{Sl^{Loe}\cdot\left[\cos(Loe\cdot\varphi) + i\sin(Loe\cdot\varphi)\right] - Sl^{Hie+2}\cdot\left\{\cos\left[(Hie+2)\varphi\right] + i\sin\left[(Hie+2)\varphi\right]\right\}}{1 - Sl^2\cdot\left[\cos(2\varphi) + i\sin(2\varphi)\right]}$$

$$= \frac{\left\{Sl^{Lo}\cdot\cos(Lo\cdot\varphi) - Sl^{Hi+1}\cdot\cos\left[(Hi+1)\varphi\right]\right\} + i\left\{Sl^{Lo}\cdot\sin(Lo\cdot\varphi) - Sl^{Hi+1}\cdot\sin\left[(Hi+1)\varphi\right]\right\}}{(1 - Sl\cdot\cos\varphi) + i(-Sl\cdot\sin\varphi)}$$

$$- Ef\,\frac{\left\{Sl^{Loe}\cdot\cos(Loe\cdot\varphi) - Sl^{Hie+2}\cdot\cos\left[(Hie+2)\varphi\right]\right\} + i\left\{Sl^{Loe}\cdot\sin(Loe\cdot\varphi) - Sl^{Hie+2}\cdot\sin\left[(Hie+2)\varphi\right]\right\}}{\left[1 - Sl^2\cdot\cos(2\varphi)\right] + i\left[-Sl^2\cdot\sin(2\varphi)\right]}$$

The formula of complex number division:

$$\frac{z_1}{z_2} = \frac{a+ib}{c+id} = \frac{(a+ib)(c-id)}{(c+id)(c-id)}$$

$$= \frac{(ac+bd)+i(bc-ad)}{(c^2+d^2)+i(cd-cd)}$$

$$= (\frac{ac+bd}{c^2+d^2}) + i(\frac{bc-ad}{c^2+d^2})$$

Here, symbols *a* to *h* are used to represent the following values.

$$a = Sl^{Lo} \cdot \cos(Lo \cdot \varphi) - Sl^{Hi+1} \cdot \cos\left[(Hi+1)\varphi\right]$$

$$b = Sl^{Lo} \cdot \sin(Lo \cdot \varphi) - Sl^{Hi+1} \cdot \sin\left[(Hi+1)\varphi\right]$$

$$c = 1 - Sl \cdot \cos\varphi$$

$$d = -Sl \cdot \sin\varphi$$

$$e = Ef \cdot \left\{Sl^{Loe} \cdot \cos(Loe \cdot \varphi) - Sl^{Hie+2} \cdot \cos\left[(Hie+2)\varphi\right]\right\}$$

$$f = Ef \cdot i\left\{Sl^{Loe} \cdot \sin(Loe \cdot \varphi) - Sl^{Hie+2} \cdot \sin\left[(Hie+2)\varphi\right]\right\}$$

$$g = 1 - Sl^2 \cdot \cos(2\varphi)$$

$$h = -Sl^2 \cdot \sin(2\varphi)$$

The output value as a real number is obtained with the following expression.

$$\textit{The output value as a real number} = \left(\left[\frac{bc-ad}{c^2+d^2}\right] - \left[\frac{fg-eh}{g^2+h^2}\right]\right)$$

Now, the amplitude normalization unit is added to this expression. In general, amplitude value of the aforementioned formula (I call it *AmpFactor* here) is derived by the following calculation.

$$AmpFactor = \frac{Sl^{Lo} - Sl^{Hi+1}}{1 - Sl} - \frac{Ef\,(Sl^{Loe} - Sl^{Hie+2})}{1 - Sl^2}$$

However, there is a problem when $Sl = 1$ (which means flat spectrum) because it makes denominators of the above calculation result in 0. Thus, the following alternative calculation method is used in my implementation when slope value is close to 1 (to be precise, $0.99 < Sl < 1.0$).

$$AmpFactor = numHarmonics - Ef \cdot numEvenHarmonics$$

$$numEvenHarmonics \text{ (the total number of even harmonics) } = \frac{Hie - Loe}{2} + 1$$

The output value of the amplitude-normalized signal (as a real number) is calculated by the following expression.

$$z = \left[\left(\frac{bc-ad}{c^2+d^2}\right) - \left(\frac{fg-eh}{g^2+h^2}\right)\right] \cdot \frac{1}{AmpFactor}$$

The explanation given so far is the basis for my implementation of Band-Limited Oscillator UGen (BLOsc). Moreover, the C++ source code is included in the CD accompanying this discussion.

Improvement: modulatable low and high harmonic indices

After making the first version of Band-Limited Oscillator UGen, I used this oscillator to help produce a composition known as *Colour Composition 2*. During the compositional process, I was frustrated with the fact that arguments *loHarmonics* and *numHarmonics* only accept integer values. This is because of the mathematical equation that band-limited oscillator is based on. It also means continuous changes (modulation) of those values are impossible. Since the lowest harmonic index and the number of harmonics have significant impacts on resultant sounds, I decided to make an improvement to my model in order to accept real numbers as possible sources for those arguments.

In the newer version of Band-Limited oscillator, the argument *numHarmonics* (the total number of harmonics), which appeared in the first version of Band-Limited Oscillator, is replaced with *hiHarmonics* (the highest harmonic index). This reorganizing of parameters makes it easier to grasp the sonic behavior, in case a composer wants to modulate both the lowest and the highest harmonic indices at the same time.

In the new model, if a floating point value is given to either *loHarmonics* or *hiHarmonics*, the interpolation occurs between two different spectra to which integer values are given. The bottom graph of figure 2-3-8 shows the case of which *loHarmonics* = 2.3. It produces an interpolated spectrum that falls between the spectrum of *loHarmonics* = 2.0 and the spectrum of *loHarmonics* = 3.0.

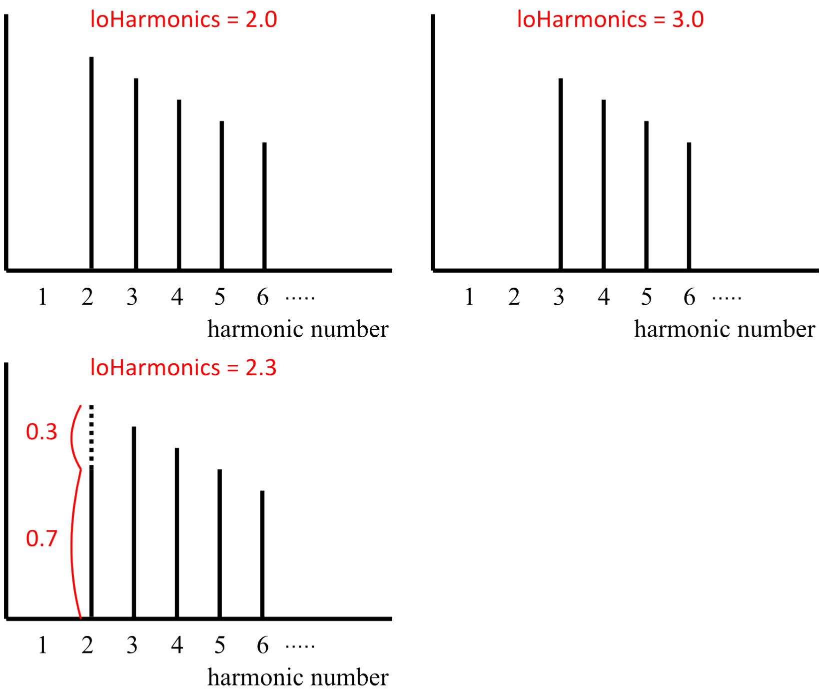


Figure 2-3-8: Floating point *loHarmonics* value

In comparison with the case of which *loHarmonics* = 2.0, the amplitude value of harmonic number 2 is multiplied by 0.7, namely 30% of attenuation. The floating point *hiHarmonics* value also works in the similar way. This is made clear in figure 2-3-9, with the use of *hiHarmonics* being equivalent to 6.4. Here the spectrum contains a harmonic number of 7, however, its overall amplitude was then multiplied by 0.4, and this should be seen in comparison with the case then the *hiHarmonics* were only equal to 7.0.

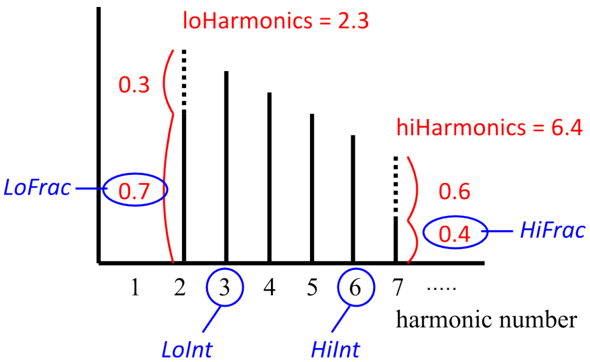


Figure 2-3-9: Floating point loHarmonics and hiHarmonics value

Based on this interpolation model, the new SuperCollider UGen called BLOsc2 is designed. To explain the calculation scheme, the following symbols are newly used:

$LoInt$ = the smallest integer value not less than Lo (ceil(Lo) in C++)

$HiInt$ = the largest integer value not greater than Hi (floor(Hi) in C++)

$LoFrac = LoInt - Lo$

$HiFrac = Hi - HiInt$

$LoAmp$ = the amplitude of the lowest harmonic index

$HiAmp$ = the amplitude of the highest harmonic index

Lo and *Hi* are given as arguments (*loHarmonics* and *hiHarmonics*) and can be either integer or floating-point value.

LoAmp and *HiAmp* need to be calculated in different ways depending on whether *LoInt* and *HiInt* are even number or odd number.

if $LoInt$ is even:

$LoAmp = LoFrac$

if $LoInt$ is odd:

$LoAmp = LoFrac \cdot evenOddRatio$

if $HiInt$ is even:

$HiAmp = HiFrac$

if $HiInt$ is odd:

$HiAmp = HiFrac \cdot evenOddRatio$

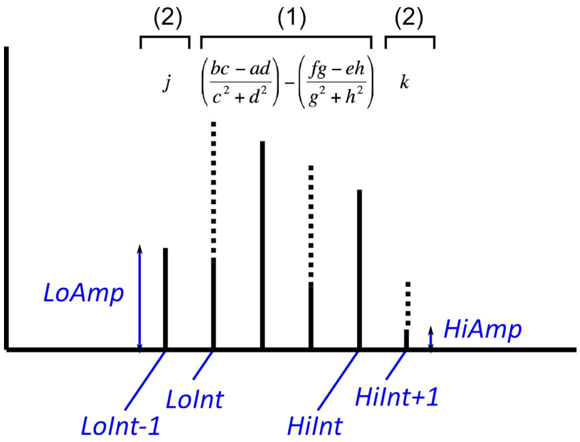


Figure 2-3-10: Calculation scheme of BLOsc2

Figure 2-3-10 is shown to explain the strategy for calculating the output signal of BLOsc2:

(1) First, derive the signal composed of partials from harmonic numbers *LoInt* to *HiInt.* For this purpose, the calculation method used in the first version of BLOsc is applied.

(2) Then, add partials of the lowest harmonic index and the highest harmonic index to the signal calculated in (1). The lowest and the highest partials can be simply understood as two sine waves.

Here, symbols *a* to *k* are used to represent the following values.

$$a = Sl^{LoInt} \cdot \cos(LoInt \cdot \varphi) - Sl^{HiInt+1} \cdot \cos[(HiInt+1)\varphi]$$

$$b = Sl^{LoInt} \cdot \sin(LoInt \cdot \varphi) - Sl^{HiInt+1} \cdot \sin[(HiInt+1)\varphi]$$

$$c = 1 - Sl \cdot \cos\varphi$$

$$d = -Sl \cdot \sin\varphi$$

$$e = Ef \cdot \left\{Sl^{Loe} \cdot \cos(Loe \cdot \varphi) - Sl^{Hie+2} \cdot \cos[(Hie+2)\varphi]\right\}$$

$$f = Ef \cdot \left\{Sl^{Loe} \cdot \sin(Loe \cdot \varphi) - Sl^{Hie+2} \cdot \sin[(Hie+2)\varphi]\right\}$$

$$g = 1 - Sl^2 \cdot \cos(2\varphi)$$

$$h = -Sl^2 \cdot \sin(2\varphi)$$

$$j = LoAmp \cdot Sl^{LoInt-1} \cdot \sin[(LoInt-1)\varphi]$$

$$k = HiAmp \cdot Sl^{HiInt+1} \cdot \sin[(HiInt+1)\varphi]$$

The newly used symbol *j* represents the sine wave of the lowest partial (harmonic number *LoInt*-1), and *k* represents the sine wave of the highest partial (harmonic number *HiInt*+1).

The output value as a real number can be derived by this formula:

$$z = \left[\left(\frac{bc-ad}{c^2+d^2}\right) - \left(\frac{fg-eh}{g^2+h^2}\right) + j + k\right] \cdot \frac{1}{AmpFactor}$$

To calculate *AmpFactor* for this newer version of Band-Limited Oscillator, the following formulae are used. Again, different schemes are applied depending on whether *Sl* (slope) is close to 1 or not.

If $Sl \le 0.99$ or $1.01 \le Sl$:

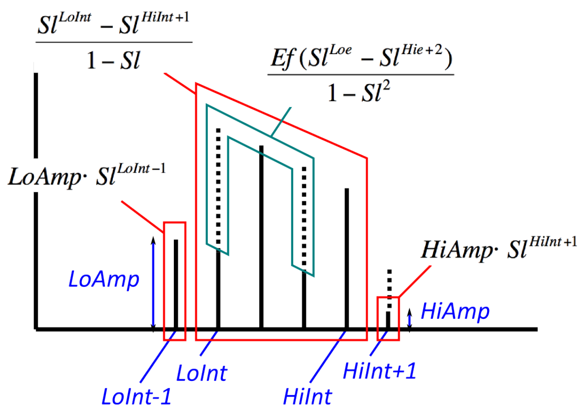
$$AmpFactor = \frac{Sl^{LoInt} - Sl^{HiInt+1}}{1 - Sl} - \frac{Ef\,(Sl^{Loe} - Sl^{Hie+2})}{1 - Sl^2} + LoAmp \cdot Sl^{LoInt-1} + HiAmp \cdot Sl^{HiInt+1}$$

If $0.99 < Sl < 1.01$:

$$AmpFactor = HiInt - LoInt + 1 - Ef \cdot numEvenHarmonics + LoAmp + HiAmp$$

where $numEvenHarmonics = \dfrac{Hie - Loe}{2} + 1$

$0.99 < Sl < 1.01$



$Sl \le 0.99$ or $1.01 \le Sl$

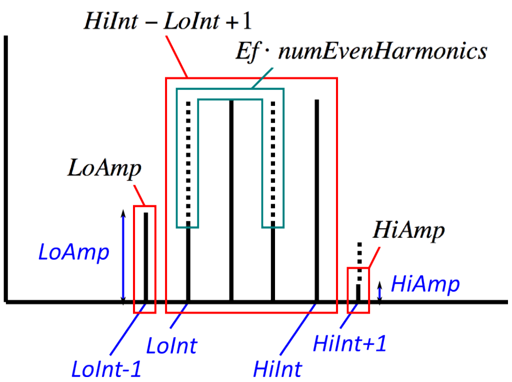


Figure 2-3-11: Calculation scheme of *AmpFactor*

In my C++ source code, the following variable names are used different from the symbols used in the above explanation.
- *Lo* is called "loHarmonics" and *Hi* is called "hiHarmonics".
- *LoInt* is called "loHarmonicsInt" and *HiInt* is called "hiHarmonicsInt" .
- *LoFrac* is called "loHarmonicsFrac" and *HiFrac* is called "hiHarmonicsFrac" .
- *LoAmp* is called "fundamentalAdjust" and *HiAmp* is called "extraHarmonicsAdjust .
- *j* is called "fractionalFundamental" and *k* is called "fractionalExtraHarmonics" .

2-4 Implementing fluctuating timbre

In this section, patches to demonstrate the basic features of Band-Limited Oscillator and patches to achieve fluctuating timbre with it are introduced. All the code examples explained below are contained in the companion CD (in a folder named "4 SuperCollider Code Examples").

Basic features and modulations of Band-Limited Oscillator

EXAMPLES 1-4 demonstrate basic functions and modulations of UGens BLOsc and BLOsc2 with GUI.

EXAMPLE 1: BLOsc demonstration with GUI
EXAMPLE 2: BLOsc modulation with GUI (Modulator: SinOsc)

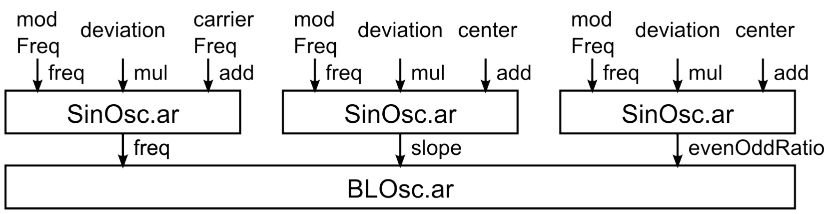


Figure 2-4-1: Patch used in EXAMPLE 2

EXAMPLE 3: BLOsc2 demonstration with GUI
EXAMPLE 4: BLOsc2 modulation with GUI (Modulator: SinOsc)

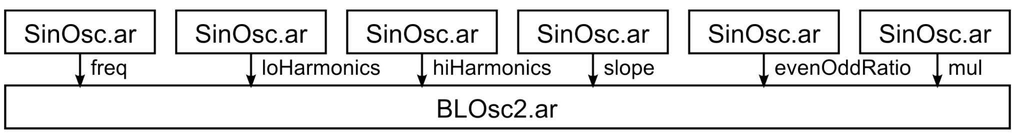


Figure 2-4-2: Patch used in EXAMPLE 4

BLOsc and BLOsc2 can also be used as a modulator. In EXAMPLE 5, three BLOsc units modulate parameters of another BLOsc.

EXAMPLE 5: BLOsc modulates another BLOsc with GUI

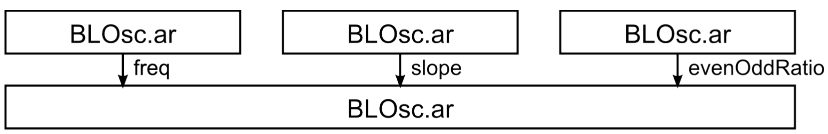


Figure 2-4-3: Patch used in EXAMPLE 5

BLOsc sequence of short sounds

Example 6 makes a sequence of short BLOsc2 sounds with percussive envelopes.

EXAMPLE 6: Colorful sequence of BLOsc2 sound with percussive envelope

In this looping pitch sequence, every short note has a different timbre resulting from a different parameter setting of BLOsc2. *loHarmonics* is controlled by exponential distribution function while the number of harmonics is fixed (*hiHarmonics* value = *loHarmonics* + 15). *slope* and *evenOddRatio* are changed based on Brownian motion.

As explained in chapter 1-7, timbre of acoustic instruments is not invariant under loudness change. To roughly simulate this phenomenon, a relationship between the spectral shape and the amplitude is introduced in this example. When BLOsc2 makes brighter timbre with higher *slope* value, the amplitude becomes louder. A relationship between *slope* value and release time is also implemented.

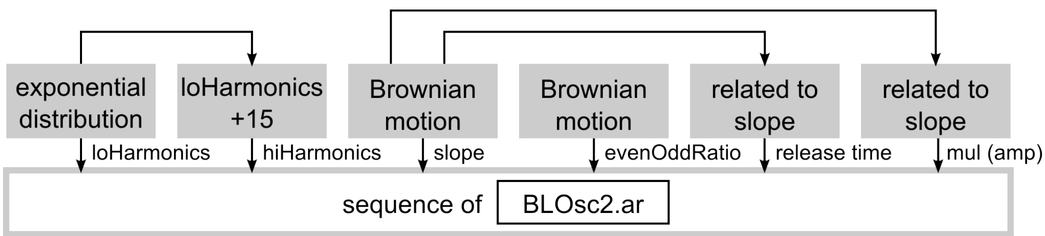


Figure 2-4-4: Pattern used in EXAMPLE 6

Fluctuating timbre in sustained sound

Next, the implementations of fluctuating timbre in sustained notes are explained. The basic approach is to apply envelopes (control signals) to parameters of Band-Limited Oscillator. In the following examples, two different methods are used to obtain envelopes. The first method is to make envelopes with noise generators. The other method is to extract envelopes from samples.

The first method uses SuperCollider's noise generator UGens such as LFNoise and LFGauss. The advantage of this approach is that a composer can have a precise control over fluctuation ranges of parameters and the frequency of fluctuation.

The second method enables a composer to obtain lively sonic fluctuations different from behaviors obtained by the first method. Field recording materials often contain background noises and unexpected amplitude changes. These elements give constantly fluctuating envelopes and occasionally produce surprising amplitude jumps, which are hard to be made with noise generators.

EXAMPLE 7: Parameters of BLOsc are modulated by envelopes made with one noise generator

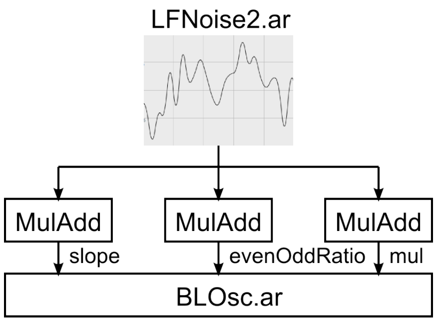


Figure 2-4-5: Patch used in EXAMPLE 7

SuperCollider’s noise generators include LFNoise0 (Step noise), LFNoise1 (Ramp noise), and LFNoise2 (Quadratic noise). LFNoise1 and LFNoise2 are efficient to make smooth envelopes suitable for implementing fluctuating timbres in many cases. LFNoise0 causes abrupt changes of amplitude, so it may be used in limited situations.

EXAMPLE 8: Parameters of BLOsc are modulated by envelopes taken from one sample

An amplitude envelope extracted from a sample controls parameters of Band Limited Oscillator. In this patch, the changes of *amplitude*, *slope*, and *evenOddRatio* occur in sync. This gives the feeling of ordered timbral fluctuation compared with examples explained later.

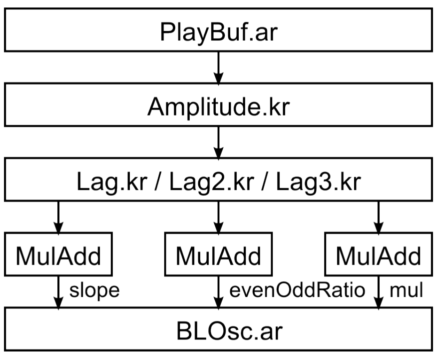


Figure 2-4-6: Patch used in EXAMPLE 8

Amplitude.kr works as an envelope follower (amplitude demodulator). Lag/Lag2/Lag3 smoothes out the extracted envelope. One of those UGens should be chosen depending on how much smoothness is preferred. LagUD/Lag2UD/Lag3UD can be used instead of Lag/Lag2/Lag3 in case different lag times are preferred for rising signal and falling signal. MulAdd values need to be adjusted depending on the character of sample and the choice of lag time (Lag/Lag2/Lag3).

EXAMPLE 9: Parameters of BLOsc are modulated by envelopes taken from three different samples

In this example, amplitude envelopes extracted from multiple samples control different parameters of Band-Limited Oscillator. This gives more unexpected and disordered feeling in timbral fluctuation compared with the previous patch.

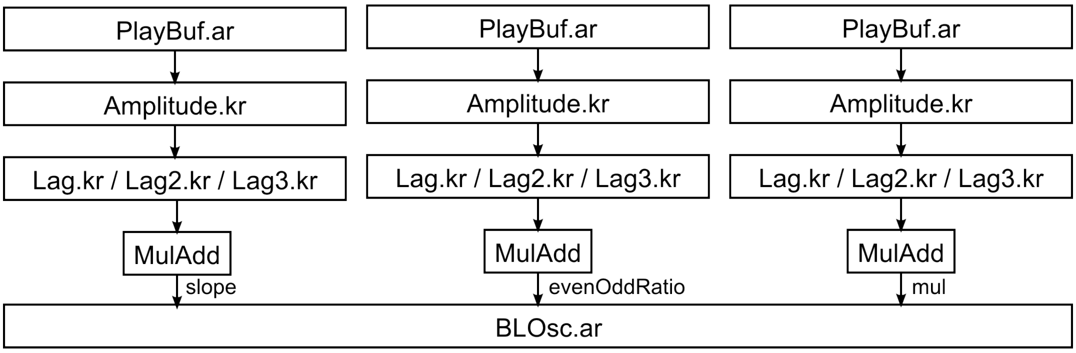


Figure 2-4-7: Patch used in EXAMPLE 9

Making harmony with long sounds with fluctuating timbre

This "fluctuating sustained sound" becomes much more interesting when it is played in harmony. Depending on patching, different degrees of synchronization among timbral fluctuations of different notes are made.

Fluctuations of all parameters and all notes are in sync

In the following example, one amplitude envelope is extracted from a sample and applied to parameters (*amplitude*, *slope*, and *evenOddRatio*) of multiple notes.

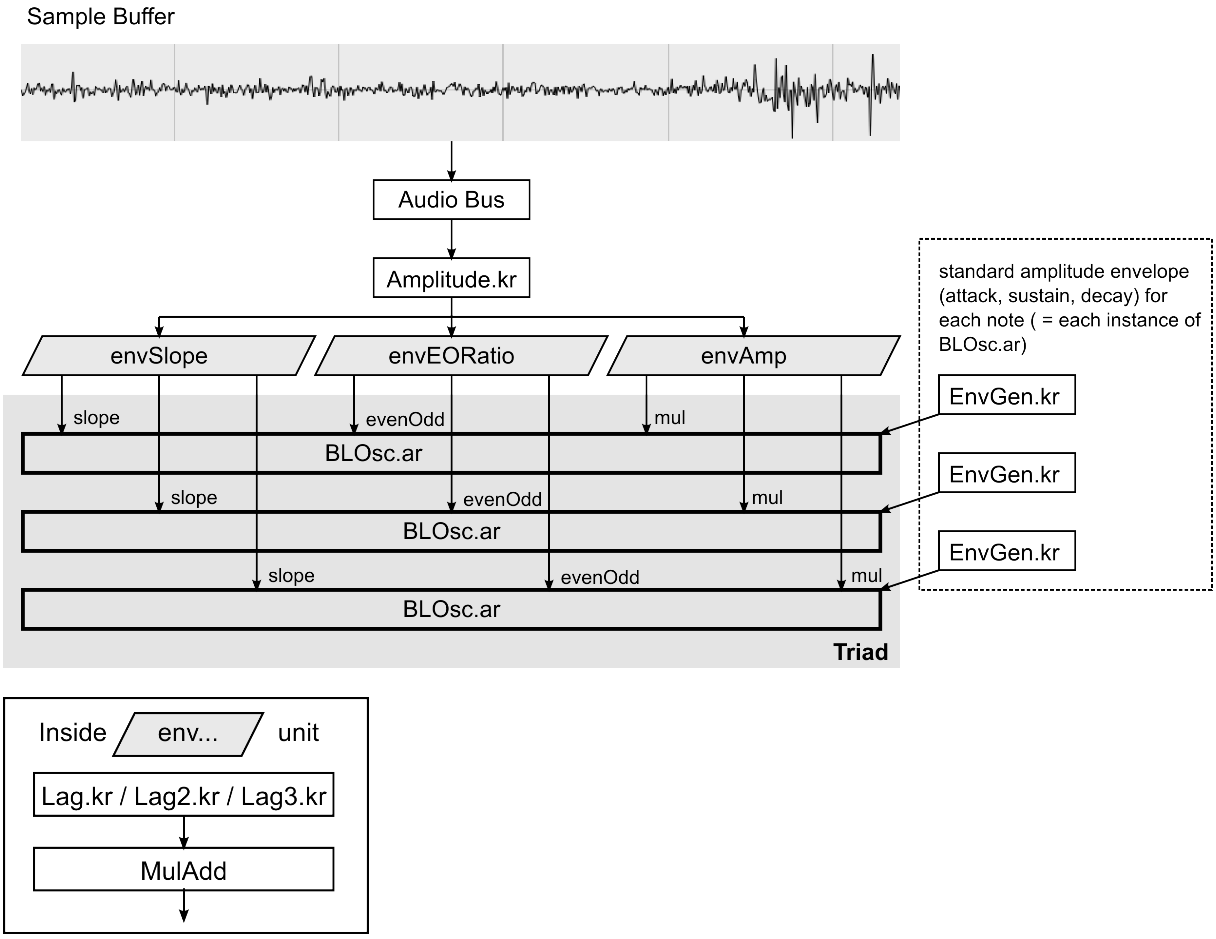


Figure 2-4-8: Patch used in EXAMPLE 10-13

EXAMPLE 10: Triad 1 (Same lagtime and MulAdds value for all notes)
EXAMPLE 11: Triad 2 (Different lagtime and MulAdds values for different notes)

The lag time (Lag/Lag2/Lag3) and MulAdd can be either same or different for each note. If different values are used for different notes, each note shows slightly different behavior in terms of timbral fluctuation.

EXAMPLE 12: 12 tone harmonic sequence 1 (Same lagtime and MulAdds value for all notes)
EXAMPLE 13: 12 tone harmonic sequence 2 (Different lagtime and MulAdds values for different notes)

In EXAMPLES 12 and 13, each of 12 notes in one octave is played at once in a random order. There is an overlap between notes.

Fluctuations of same parameters of different notes are in sync

In the following two examples, different envelopes are applied to different parameters. However, fluctuations of same parameters of multiple notes are in sync.

Instead of preparing different samples for different envelopes, multiple amplitude envelopes are extracted from one sample by changing the reading positions (with random start time) for an efficiency reason. Playback speeds and loop lengths of sample are also randomized for different envelopes.

Compared with EXAMPLE 10-13, the following patches generate more frequent timbral fluctuation because different parameters exhibit changes at different moments.

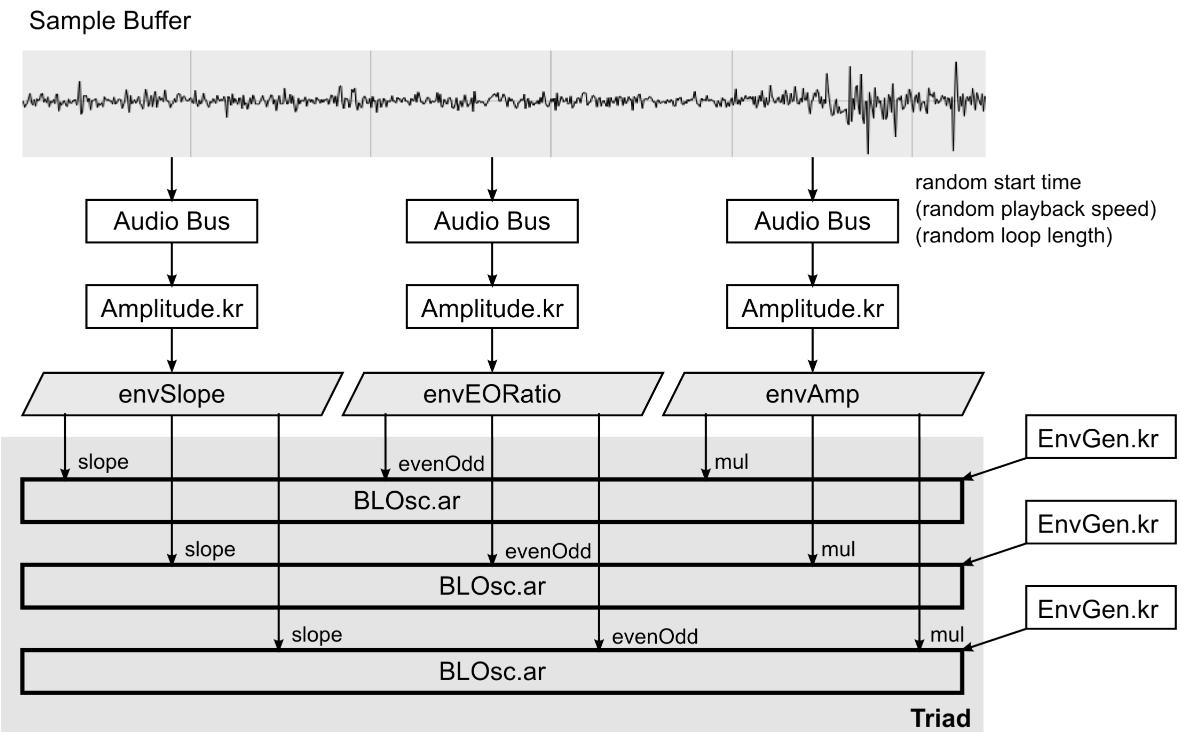


Figure 2-4-9: Patch used in EXAMPLE 14-15

EXAMPLE 14: Triad
EXAMPLE 15: 12 tone harmonic sequence

Fluctuations of different parameters of same note are in sync

In the following two examples, different envelopes are applied to different notes. However, fluctuations of multiple BLOsc parameters in one note are in sync because they share same Amplitude.kr.

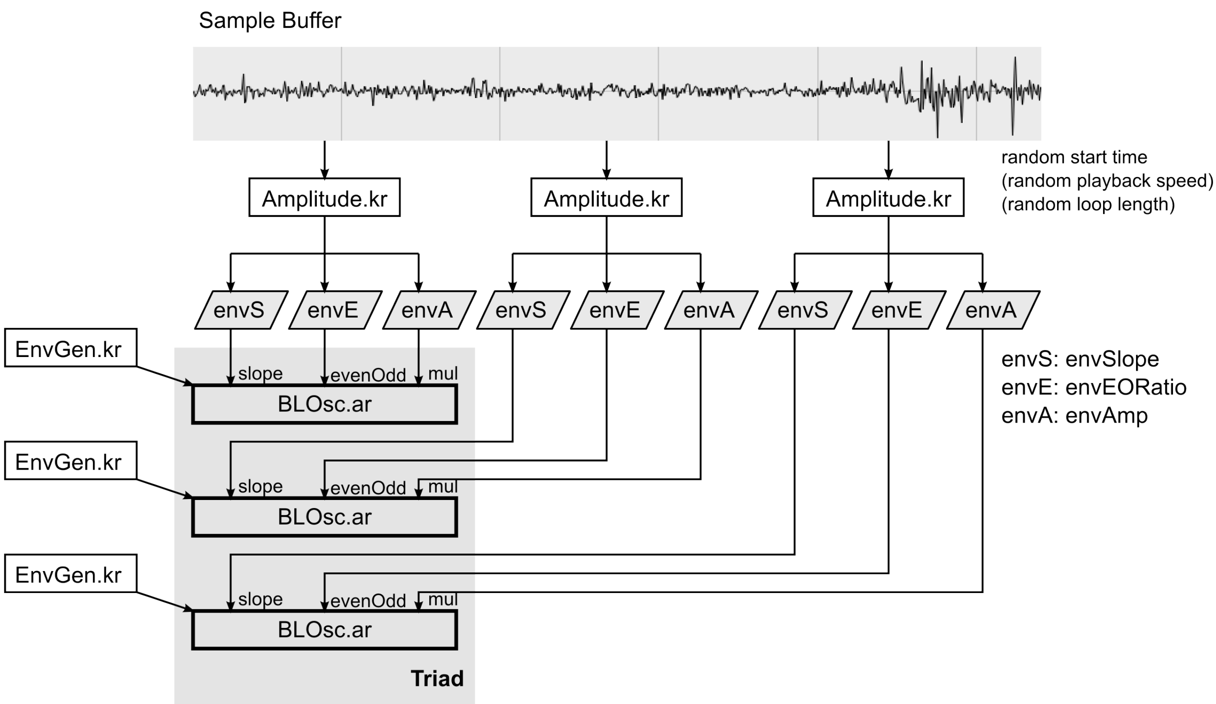


Figure 2-4-10: Patch used in EXAMPLE 16-17

EXAMPLE 16: Triad
EXAMPLE 17: 12 tone harmonic sequence

None of fluctuations are in sync

In the following two examples, different envelopes are applied to different parameters and different notes. The lack of synchronization with other parameters and other notes gives disordered timbral fluctuation as a whole.

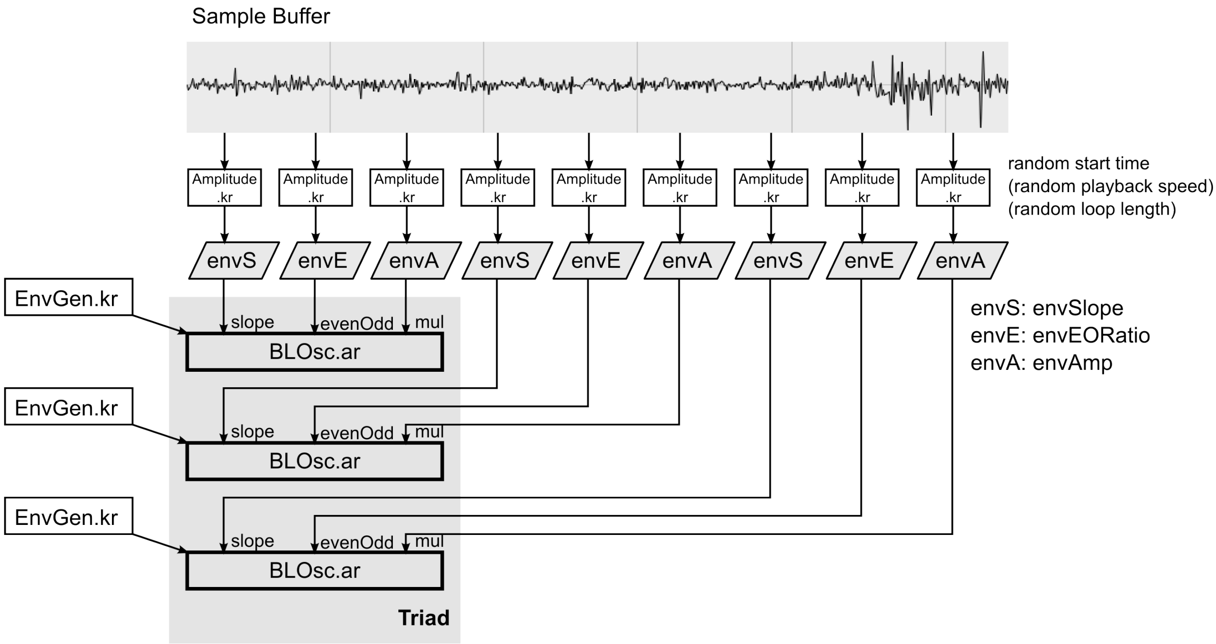


Figure 2-4-11: Patch used in EXAMPLE 18-19

EXAMPLE 18: Triad
EXAMPLE 19: 12 tone harmonic sequence

Changing the fluctuation range over time

The aforementioned techniques for making fluctuating timbre become more useful in practical compositional situations by introducing a function to change the fluctuation range over time. For example, a composer might want to avoid too obvious timbral fluctuation in the beginning of a piece and want much bigger timbral fluctuation in the middle part or coda. This can be achieved by introducing other envelopes that control fluctuation ranges of parameters.

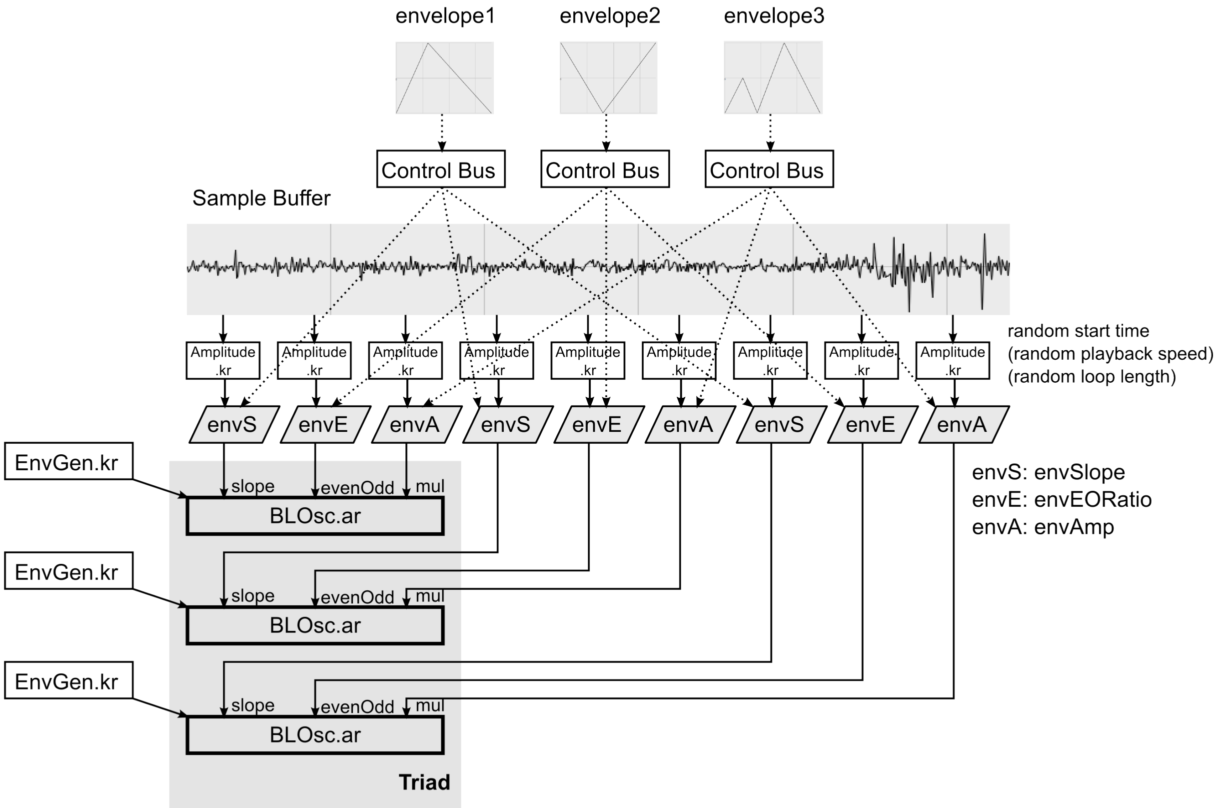


Figure 2-4-12: Patch used in EXAMPLE 20

In EXAMPLE 20, the timbral fluctuation is minimum in the beginning and reaches maximum about 60 seconds later. Then, the fluctuation becomes smaller again.

EXAMPLE 20:   12 tone harmonic sequence

Using BLOsc2 instead of BLOsc

The next example uses BLOsc2 instead of BLOsc. By modulating *loHarmonics* and *hiHarmonics* parameters, BLOsc2 produce more dramatic timbral changes than BLOsc. In EXAMPLE 21, envelopes made with five different noise generators (LFNoise2) modulate parameters of one instance of BLOsc2.

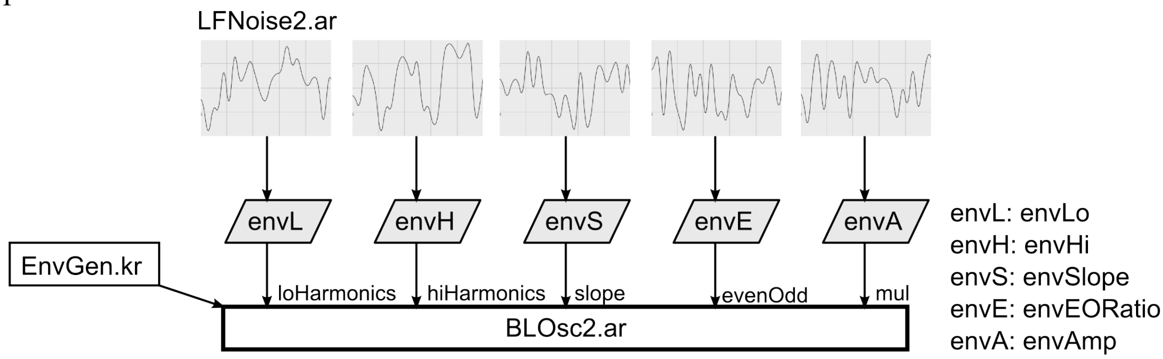


Figure 2-4-13: SynthDef for EXAMPLE 21

EXAMPLE 21:   12 tone harmonic sequence

2-5 Sound transformations with band-limited oscillator

Band-Limited Oscillator can be used not only for sound generation but also for sound transformation. In this section, convolution techniques using BLOsc2 for transformation purpose is explained.

Convolving a sampled sound and BLOsc2

The patch shown in Figure 2-5-1 performs the convolution of a sampled sound and a sound made with Band-Limited Oscillator.

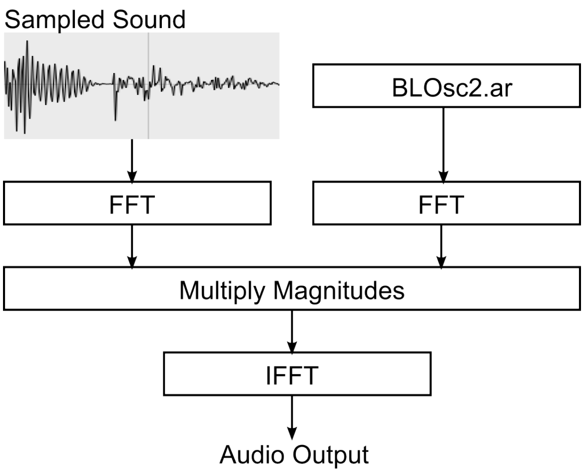


Figure 2-5-1: Convolving sampled sound and BLOsc2

This patch can be considered as a type of brick-wall filter. In the resynthesis process, the frequency content of sampled sound only within the band specified by BLOsc2 is produced. *loHarmonics* and *hiHarmonics* of BLOsc2 determine the cut-off frequencies of this brick-wall filter. The advantage of this system is that the filter response can be changed over time by moving parameters of BLOsc2. By narrowing the band (increasing *loHarmonics* value and/or decreasing *hiHarmonics* value), the sampled material starts obtaining a resonant and synthetic quality. Thus, this system is useful for morphing a concrete sound into a sound with synthetic quality. It can also be utilized to transform a noisy sound into a sound with clear pitch and vice versa. Lowering *slope* value emphasizes the lower frequency components of the material. Lowering *evenOddRatio* creates artifacts with a metallic quality in the resultant sound.

EXAMPLES 22 and 23 of SuperCollider code show this convolution process.

EXAMPLE 22: First Input: BLOsc2, Second Input: Sample Buffer
EXAMPLE 23: First Input: Sample Buffer, Second Input: BLOsc2

In these patches, phase vocoding UGen PV_Magmul is used. PV_Magmul multiplies the magnitudes of two buffers of FFT data but uses the phase values of the first input. This means that the order of the inputs has an influence on the resultant sound. Depending on the material that is convolved with the BLOsc2 sound, the efficient order is different. I found that

EXAMPLE 22 is good for harmonic materials and EXAMPLE 23 is good for noisy sounds and field recording materials, though it is of course not the absolute rule.

Convolving noise and BLOsc2 (Implementation of band-limited noise)

By using white noise as an input instead of a sampled sound, variable band-limited noise can be implemented. Because of the steep cut-offs that band-limited oscillator produces, the change of *loHarmonics* and *hiHarmonics* of BLOsc2 in this band-limited noise implementation generates audible pitch movements, which is less obvious in standard filters.

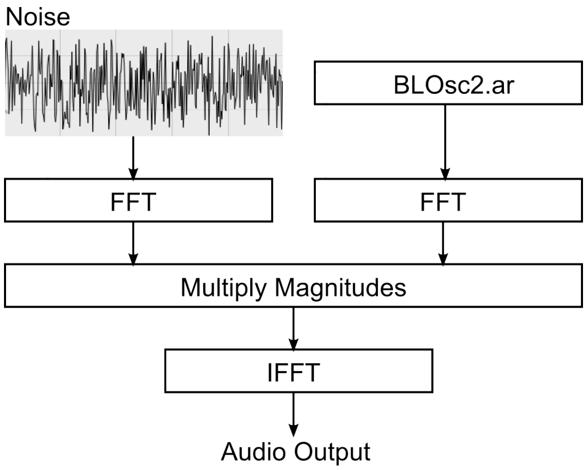


Figure 2-5-2: Convolving noise and BLOsc2 (implementation of band-limited noise)

EXAMPLE 24: First Input: White Noise, Second Input: BLOsc2

By replacing white noise with different types of noise, more variations of band-limited noise can be obtained. Using low frequency noise and changing its frequency can alter the quality of band-limited noise over time. Using random impulse (Dust or Dust2 in SuperCollider) and changing its density produces the different quality of textural change.

Convolving a sampled sound that has a noisy quality with a sound from a noise generator also makes the variations of noise. (Figure 2-5-3)

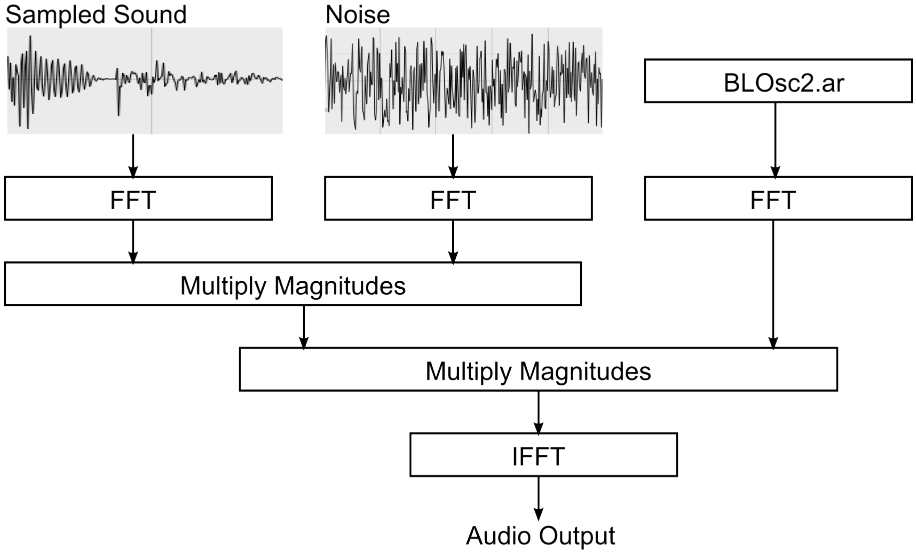


Figure 2-5-3: Making variations of noise by convolving a sampled sound and a sound from a noise generator

EXAMPLE 25: First Input: Sample * White Noise, Second Input: BLOsc2

Making unique spectral shapes with multiple BLOsc2 units

It is also possible to make band-limited noises with unique spectral shapes by convolving a noise with a sound made by adding multiple BLOsc2 sounds. (Figure 2-5-3, Figure 2-5-4)

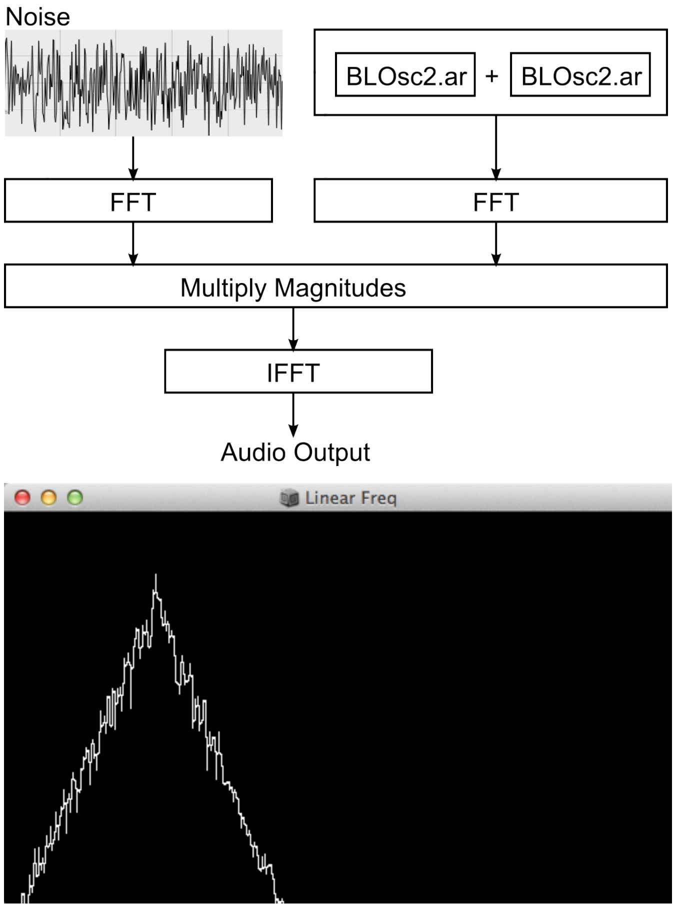


Figure 2-5-4: White Noise is one input. The addition of two BLOsc2 sounds is another input. (top) Spectrum made with this patch with the parameter setting of EXAMPLE 26. (bottom) The spectral shape becomes different depending on the parameter settings of BLOsc2 units.

EXAMPLE 26: First Input: White Noise, Second Input: BLOsc2 + BLOsc2

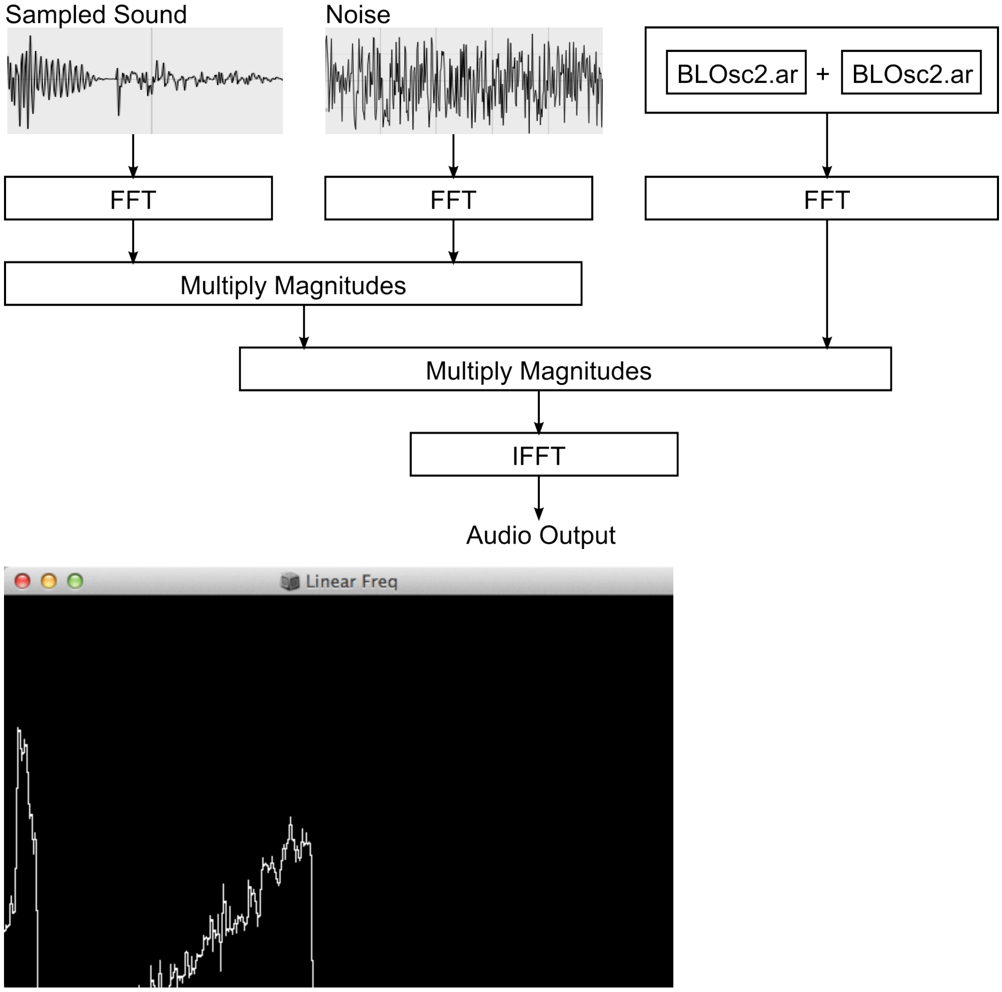


Figure 2-5-5: Convolved sound of a sample and white noise is one input. The addition of two BLOsc2 sounds is another input. (top)
Spectrum made with this patch with the parameter setting of EXAMPLE 27. (bottom)
The spectral shape becomes different depending on the parameter settings of BLOsc2 units.

EXAMPLE 27: First Input: Sample * White Noise, Second Input: BLOsc2 + BLOsc2

Chapter 3 Timbre Composition Techniques

Up to previous chapter, the technical means to achieve timbral movement have been explained. In this chapter, compositional theories and techniques regarding timbral movements are examined. The followings are some of the important keywords covered in this chapter: time scale, system, transformation, consistency, complexity, and layer.

3-1 Two aspects of "Composing Sound"

As it has was previously mentioned in chapter 1, the concept of "composing sound" has personally implied an inclusion of the following two aspects:
(1) Creating new and original sounds
(2) Realize new sound movements

In connection to the above I would like to start explaining how my own compositional ideas, regarding timbre, have been connected to these two facets of composing sound.

(1) Creating new and original sounds
This simply means making new sounds that has have not previously existed. This act is analogous to the act of a painter who creates new colour by blending different pigments on a palette. Historically, the attempt and fascination to make new sounds was tried before the birth of electronic music. In the 20th century this can be seen in the work of composers like Alban Berg who blended many instruments and created a single perceptual sound colour in the beginning notes of the third song of *Altenberg Lieder*. (Erickson, 1975)

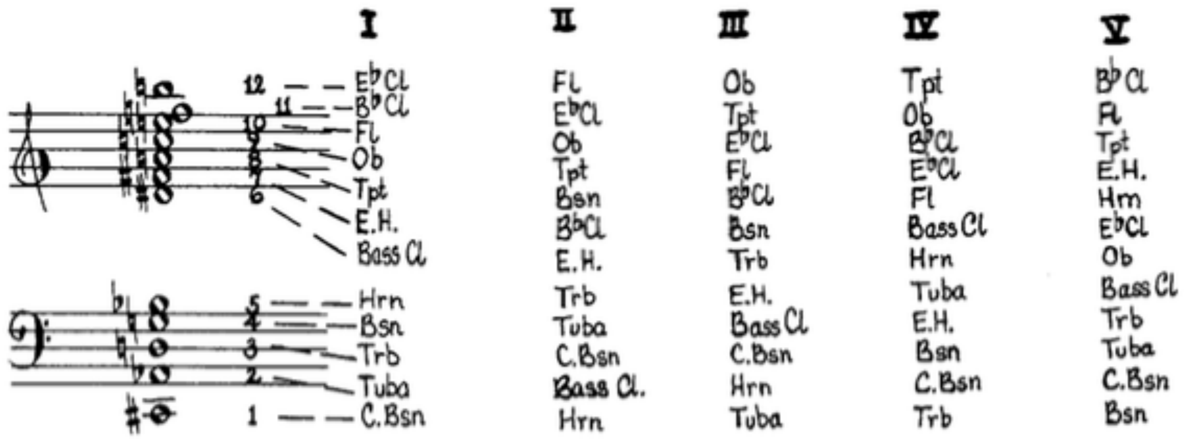


Figure 3-1: The instrumentation of first five sounds of the third song of Berg's *Altenberg Lieder* (Erickson, 1975, p. 168)

However, electronic music of course has much more potential for making new sounds by means of synthesis or processing. Technically speaking, this act can be considered as composing in microtime level, or composing waveform. In the early days of electronic music, composing waveform was done by simply adding sine tones. Later, the phrase "composing sound" tends to imply manipulation of individual samples and non-standard synthesis. Non standard synthesis, such as Gendy, SSP or ASP, often requires calculations of individual sample values with some random functions and it tends to result in noisy qualities. However the Band- Limited Oscillator I have developed is very different from existing non-standard synthesis because it does not use any random function and the generated sound is completely pitched. On the other hand, this synthesis method aims at making arbitrary spectra, which means arbitrary waveforms in time domain interpretation, by calculating individual sample values using complex number, thus I believe it can be discussed in the context of composing sound.

In most regards, possibilities to make new sounds empower composers to add sonic variation to their music. If newly created sound has unique characteristics in itself, the result will be a certain novelty being brought to the music, which can sometimes translate into a mysterious impression upon the listeners. Personally, when I encounter a sound I am not familiar with while listening to someone else's electronic music composition, the question is often focused around how the composer was able to make this sound. I personally maintain an aesthetic preference for electronic music to instrumental music because I enjoy this mysterious feeling I often obtain from unfamiliar sounds.

It is also meaningful to synthesize a new sound whose quality falls in between two existing sounds. In this way, the summation of qualities of two sounds, creates a new sound that can become the subject for a composition. The existence of such a new sound, that connects two different materials, plays an important role in electronic music, because it contributes to both the consistency and the complexity of composition using this medium. Later I will discuss about how connecting materials in this way can be done and has had relevance to my music.

(2) Realize new sound movements

In my opinion, "composing sound" means not only making waveforms, but also making new sound movements. With this in mind, Koenig's suggestion of the concept of "composing sound" has an important and fundamental significance to issues I am also working with.

> The electronically produced sound does not have to be homogenous and continuous. In its capacity of structured field it is variable in all directions . . . In electronic music, one complete sound can consist of several individual components whose determinants may change in the next sound. In this manner the individual sound becomes data storage with references both to itself and to the next sound. The sound is a closed unit and becomes the means of formation; each sound characteristic (the course of its pitch, loudness, timbre) becomes a formal event. Instrumental music composes "with" sounds, electronic music composes sounds. (Koenig, 1963, p. 2)

We can read from this text that Koenig took into consideration movements and changes of a sound when he talks about "composing sound". I think reinterpreting the concept of "composing sound" in the light of movement is, therefore, a meaningful artistic aim — especially for today's electronic music. Moreover, the development of computer has made many new synthesis techniques possible, allowing for far more accessible tools to manipulate all sample values and to design arbitrary waveforms. This means synthesizing brand-new (static) sounds, ones that have not been heard before, is getting more and more difficult to do. On the other hand, I believe we still have ample possibilities to make new sound movements, an area that for the most part has not yet been fully explored. The exploration of new movements will lead us to find novel musical gestures.

It should be emphasized too that sound movement is closely related to the concept of "fluctuating timbre". Specifically some roles timbral movement plays in electronic music composition would include: giving sound an overall quality of vibrancy; making its directionality part of the musical development; connecting its different materials; and controlling what material comes to foreground and what remains within the background. These key points will soon be discussed in greater detail.

3-2 Time scale and directionality of timbre movement

I would like to classify timbre movement into two types, these divisions will depend primarily time scale and the existence or non-existence of directionality. Respective of these types will also be the way movement plays a defining and differentiating role within a particular composition.

Short term, non-directional timbre movement

One example is the sound of a sustained Band Limited Oscillator, whose timbre is related to other parameters (e.g. *slope*, *evenOddRatio*) and whose output is continuously affected by low frequency noise. This may also includes a sequence of percussive sounds whose parameters are randomized at every triggering moment. This timbre fluctuation gives an overall vibrancy to the quality of sound, but it does not contribute to a large-scale formal evolution.

Long term, directional timbre movement

An example of the above includes slowly increasing cutoff frequency of filter and gradually increasing *slope* values of a Band Limited Oscillator in order to cause larger time scale changes of timbre. This type of timbre fluctuation provides some directionality to music and plays a role in the structural evolution of a composition.

In connection to this idea, Curtis Roads proposed related concepts while his discussing texture. The categories he proposed are *stationary processes* and *weighted stochastic texture.*

> Many complex musical textures resemble what statistics calls *stationary processes*. A stationary process exhibits no trend. The texture has a fixed mean value and fluctuates around the mean with a constant variance. A stationary process is not necessarily static in time, but its variations remain within certain limits, and are therefore predictable . . . To impose a trend is to gradually change the texture. This may take place over time periods associated with the meso time scale (i.e., many seconds). A trend converts a stationary texture into *weighted stochastic texture*. (Roads, 2001, p. 334-335)

An example of *stationary processes* Roads suggested is a cloud of grains scattered over a broad zone of frequencies. According to Roads, the ear perceives such change "as a deviation from the stationary". An example of *weighted stochastic texture* is the time-varying change of a filter's parameters, such as opening or closing of the bandwidth and moving the central frequency as it is applied to a cloud of *grains*.

Of course my discussion is not limited to granular synthesis but concerns any timbral movement. In addition, the *short term timbre movement* I am discussing deals with more flexible time scale than Roads's s*tationary processes*. Any specific deviations of granular synthesis generally occur on the level of milliseconds. On the other hand, *short term timbral movement,* such as I have in mind tends to take place over a few seconds, and is dependent on tempo and context, in so far that it does not need to contribute to directional musical evolution. Nonetheless, Roads's categorization and my categorization do seem to have a certain resemblance.

In connection to this I would like to take a look at examples of *short term* and *long term* timbral movement in specific compositions. In a piece by Morton Subotnick, called *Touch*, we can hear the mix of pitched sound and noise-based sound (from minute 12:14). The random appearance of noisiness endows the lively quality of sound (*short term, non-directional timbre movement)*. Taking a broad view, the proportion of noise to pitched sound changes over time, contributing to the musical evolution (*long term, directional timbral movement*).

Lumière from Bernard Parmegiani's *La Création du Monde* is another example showing both types of movements. Almost the entire piece is constituted of small sonic particles, with each particle having a different pitch, brightness, and degree of noise. This deviation of parameters makes rich texture (*short term, non-directional timbre movement*). At the same time, there is macro-time change of texture in this piece. Pitched particles are dominant in some part and other noisier particles are more present in another part. This larger scale textural change plays formative role in this piece (*long term, directional timbre movement*).

Timbre movements in some instrumental pieces can be interpreted from this perspective. In Webern's orchestration of Bach's *Ricercare* from *the Musical Offering*, timbral change happen within 8 bar cycles and can be seen as *short term, non-directional timbral movements*. Taking a look at the first 8 bars, the trombone, horn, and trumpet appear one after the other. However, the alteration of instruments, in this short section, does not clearly show which direction the music is heading. However, examining the timbral change in longer time scale, the 8 bar cycles begins with only brass instruments, then the instrumentation was taken over by woodwind. After that, it becomes the mixture of brass and woodwind. This timbral development achieved by the alternation of instrumental sections composes the bigger structure of this piece, therefore it can be considered as *long term, directional movement*.

	1	2	3
1	Trombone	Horn	Trumpet
2	Flute	Clarinet	Oboe
3	Bassclarinet	Trombone	Bassoon
4	English Horn	Horn	Bassclarinet
5	Trumpet	Oboe	Clarinet
6	Bassclarinet	Bassoon	Violoncello

Figure 3-2: The change of instruments in Webern's orchestration of Bach's *Ricercare* (Koenig, 1965b, p. 2)

I would like to mention another very different musical example. As it was earlier mentioned, I have been a fan of techno music for long time. In light of this I have seen performances of many internationally acclaimed DJs and from these experiences I have gained an impression that skilful DJs are always, even somewhat restlessly, touching a mixer during their performance. I do not however think they perform this action simply to show off their skills, in fact I believe this action is a result of responding to a certain musical necessity.

Although what they do on stage is not always completely clear, what it is apparent is that DJs are always changing timbre (*short term timbre movement*) with equalizer. This technique is done to give a vibrant quality to what is, quite often, very repetitive music. At the same time, they sometimes adjust equalizer slowly, for instance, when they move from one track to the other (*long term timbre movement*). A perceptive DJ, based on his/her experience, would therefore be well-versed in knowing might the importance of changing timbre on both these different time scales.

3-3 Algorithmic approach to timbre movement

Next, I would like to propose a method to algorithmically deal with *short term, non-directional timbre movement* and *long term, directional timbre movement*. The method I suggest is based on a serial idea and can utilize the following procedures.

1. Determining horizontal and vertical boundaries.

Horizontal boundaries indicate start and ending times of a section whereas vertical boundaries imply a maximum and minimum value that a given parameter can take. In the case of parameters related to timbre, the correct setting of boundaries is almost always important to produce sounding satisfactory results.

2. Determine the horizontal and vertical divisions between boundaries.

Horizontal division means a time grid. A very simple example is the parametric change happening at the entry moment of every bar. The setting of time grid affects listener's perception of rhythm. Vertical division means quantization unit of concerned musical parameter. In case of pitch, scales are considered as quantization unit in tonal music. In twelve tone series, one semitone corresponds to a quantization unit. Concerning timbre related parameters, it seems to be pointless to make a complex quantization system when using synthesis methods such as Band Limited Oscillator. On the other hand, some synthesis methods, such as FM, require careful quantization to obtain results a composer wishes.

3. Determine the movement (vector) of parameter.

For example, a parameter value can change linearly, randomly, or with drunken walk. Boundaries, divisions and movements can be changed over time.

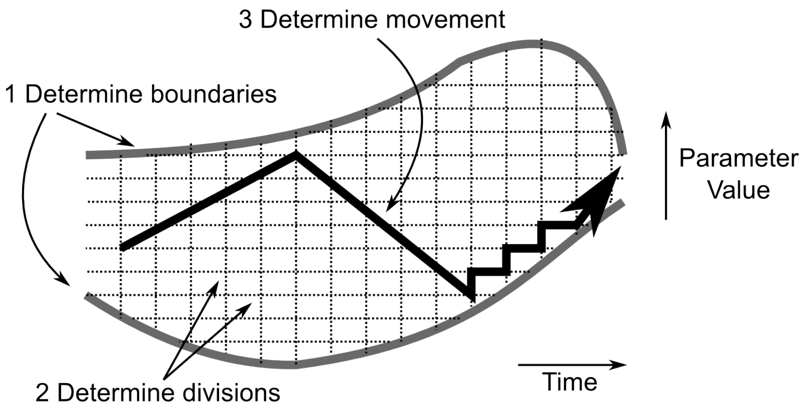


Figure 3-3-1: Procedure to choose parameter value

I would like to examine the application of this procedure to make timbral movements.

1. Boundaries

Determining time variant boundaries is same as making tendency mask. The shape of mask has a strong influence on the directionality of timbre movement.

2-a. Horizontal division (time grid)

Concerning the timing of timbral change, I would like to propose different schemes than thinking in relation to beat and meter. With this in I have categorized the methods of dividing time, primarily based on continuity and randomness.

-*continuous*: Parameter value keeps changing all the time. In other words, the time grid is very small. This kind of constant change of timbre can be easily achieved in electronic music, but difficult in instrumental music.

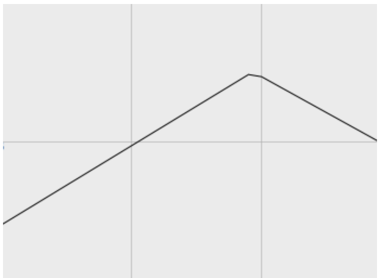


Figure 3-3-2: Continuous

-*step(stutter)*: Parameter changes occur in constant time interval, or based on some rhythm pattern. This timbre movement can be seen in some instrumental music, which intends to make quasi continuous timbral change. For instance, the alternation of instruments occurs every few notes in Bach-Webern's *Ricercare*. Before the advent of electronic music, composers might move the timbre in this way out of necessity for achieving perceptually smooth timbral change. On the other hand, this gesture has its own charm different from continuous parameter change, thus can be used in electronic music, too.

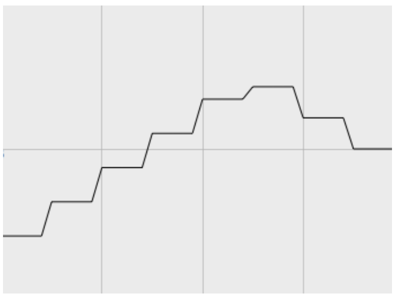


Figure 3-3-3: Step (stutter)

-*Irregular*: Different from *step*, the time interval of parameter change is randomized. This gesture gives unpredictable impression to listeners. There are intermediate states between *step* and complete *irregular* depending on the degree of randomization.


Figure 3-3-4: Irregular

Regarding *step* change, it is also important whether timbre change is synchronized to phrasing of pitch or not. In bar 85 of Debussy’s *Prélude à l'après-midi d'un faune*, the change of timbre in woodwind section occurs based on phrasing, though the overlapping string section whose note length is different from that of woodwind section makes intricate texture. (Figure 3-3-5) The beginning of Boards of Canada’s *Open the Light* is a case of (rather popular) electronic music whose timbre change synchronizes to phrasing.

Figure 3-3-5: Timbre change in bar 85 of Debussy's *Prélude à l'après-midi d'un faune*

Berlioz's "March to the Scaffold" from *Symphonie Fantastique* has the passage that can be seen as "the first bit of klangfarbenmelodie of modern times in the West". (Ericsson, 1975, p. 86) From bar 109, melody line is split into different instruments. In other words, the timbral change is not synchronized to the phasing. Bach-Webern's *Ricercare*, which have already been mentioned, is another example showing timbral change that is not synchronized with phrasing. The out-of-synchronization with a melodic motion obscures the moment of timbral change, the perceptible result then takes on a more *continuous* effect, as opposed to the case when melodic and timbral changes are synchronized. A kind of confusion is caused by this asynchronous relationship —between pitch and timbre — and this is an interesting effect, one certainly worth exploring in composition.

Figure 3-3-6: Excerpt from Berlioz's *Symphonie Fantastique*. The first pizzicato of string is in bar 109.

2-b. Vertical division (quantization unit of parameter value)

Earlier I mentioned that using a sophisticated quantization system for Band Limited Oscillator was not so fruitful. This is because the change of parameter values and the change of perceived timbre have almost linear relationships. In FM synthesis, parameter values and the resultant sound do not have linear relationship. When changing the harmonicity ratio, the result will be very different depending on the quantization unit. If the harmonicity ratio only takes integer values into consideration, harmonic sound inevitably result. However, if the quantization unit is a floating point value, then FM synthesis has the ability to generate inharmonic sounds. To summarize this point, we need to take into consideration the relationships between parameter values of concerned synthesis method and the resultant sound to determine the proper vertical division.

3. Movements

I made a list of some possible movements achieved by different algorithms (distribution functions) that can be applied to timbre related parameters.

Linear

It gives sound transformation with clear directionality.

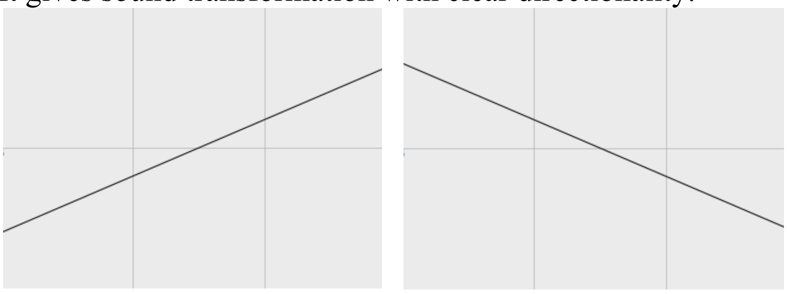


Figure 3-3-7: Linear

Random distribution (noise)

It gives no continuity and no directionality in the transformation. The combination of small time grid and random parameter movement will be resulted in *short term, non-directional* timbre fluctuation.

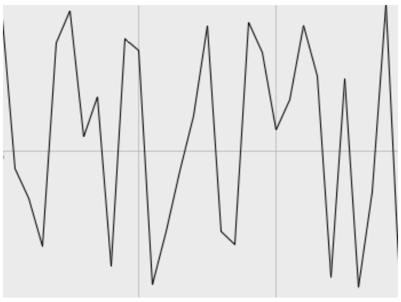


Figure 3-3-8: Random distribution (noise)

Brownian motion (random walk)
It gives both continuity and randomness to transformation.

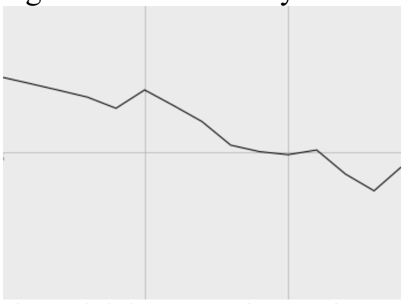


Figure 3-3-9: Brownian motion (random walk)

Beta distribution
It makes alternation between two contrasting sound qualities.

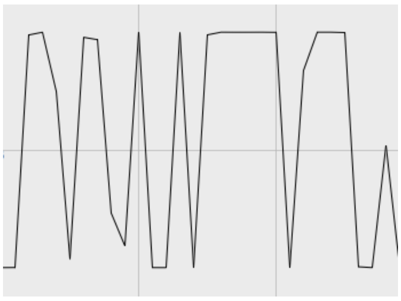


Figure 3-3-10: Beta distribution

Gaussian distribution
It makes subtle timbre fluctuation from the mean state.

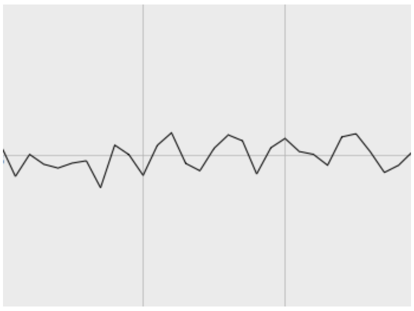


Figure 3-3-11: Gaussian distribution

Exponential distribution
It often makes subtle timbre fluctuations, but could also make an extreme jump once in a while.

Figure 3-3-12: Exponential distribution

They are just a few examples. Using other distribution functions and chaotic functions can therefore assist in making distinct and differentiated timbral motions. By combining different settings of boundaries, horizontal and vertical divisions, and movements, we can obtain various timbral gestures. Both *short term, non-directional timbre movement* and *long term, directional timbre movement* can be achieved with this procedure. The followings are some examples.

Combination1
boundaries: widening tendency mask
horizontal grid: continuous change
vertical grid: not specified
movements: random distribution

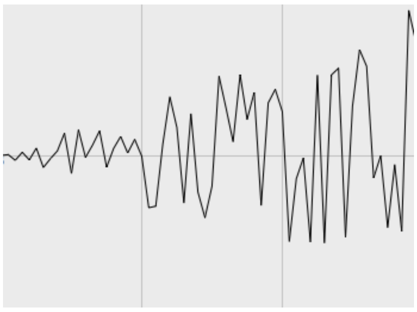


Figure 3-3-13: Combination1

Combination2
boundaries: decreasing upper and lower limit
horizontal grid: step (regular) change
vertical grid: not specified
movements: random distribution

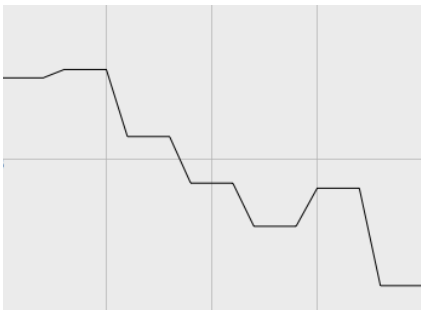


Figure 3-3-14: Combination2

Combination3
boundaries: narrowing tendency mask
horizontal grid: irregular change
vertical grid: not specified
movements: exponential distribution

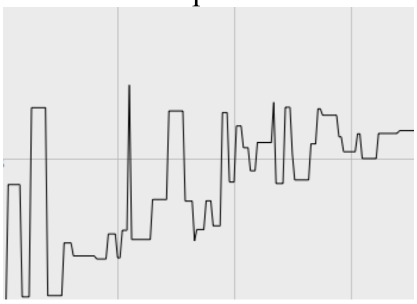


Figure 3-3-15: Combination3

3-4 Timbre movement and system

The methods explained in the previous section enable the systematic treatment of timbre movement. In this section, I would like to examine how the system of timbre movement contributes to electronic music. I start the discussion by briefly addressing why systems are important for composition in general.

3-4-1 The role of systems in electronic music

The followings are some of the roles systems play in compositional process.

1. Systems have the ability to be useful and effective tools for finding new musical gestures. There are musical gestures that can be obtained only by introducing particular system. For example, sequences obtained by twelve tone system cannot be achieved by any other way.

2. Systems enable higher-level control, especially in electronic music. There are musical ideas that can be achieved only by focusing on higher-level behavior, instead of indicating precise execution of certain parameters.

3. Systems contribute to the consistency of composition. This is regardless of whether a system is applied to pitch or time or another parameter; consistency is typically enhanced by such means, unifying different musical materials together by applying common systems. The necessity to achieve consistency is the biggest reason that I believe systems of timbre are essential to electronic music. In chapter 3-1, I wrote about making sound, which has not existed before. The possibility to obtain new sounds with various synthesis and processing methods is the potential of electronic music. On the other hand, composers need to consider carefully how to integrate different materials made with different methods into one single composition. Otherwise, musical unity will be easily lost. From this perspective a system of timbre can be considered as a substructure of a sonically rich electronic music composition, because it embodies the totality of the piece, which is composed from a variety of materials.

3-4-2 System of timbre

Musical system is interpreted as the common rule to be applied to different sounds. Scale can be considered as a system of pitch and meter can be considered as a system of time. But with this in mind, the question of what actually defines a system of timbre is raised.

System of pitch	(e.g. scale, 12 tone series)
System of harmony	(e.g. chord, cadence)
System of time	(e.g. rhythm, meter)
System of timbre?	

Figure 3-4-2-1: Systems of different musical dimensions

As mentioned in the previous section, one of the biggest roles of a system is to ensure the consistency of composition by binding various materials together. Thus, to think about a system of timbre is beneficial because it forces thoughts about ways timbre can be connected with multiple materials representing a diversity of sonic qualities. From this position, I would like to suggest the following three approaches as being relevant for connecting different materials with the overall concept of timbre.

1. Interpolate from one material to the other
2. Add common sonic quality and related movement to different materials
3. Introduce an amount of interaction between different materials

All of the above involve some movements of timbre. Thus, a "system of timbre", from my perspective actually suggests a "system of timbral movement". Elaborating on this in greater detail requires each of the aforementioned approaches to be studied in closer detail.

1. Interpolation

Interpolation, or smooth transition from one sound to the other, is done by filling the gap between two distant states by introducing mediating states. Interpolation is attempted in instrumental music to certain extent.

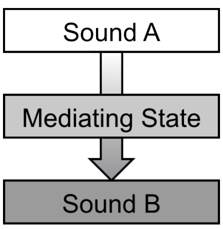


Figure 3-4-2-2: Interpolation between materials

It is impossible to make continuous interpolation between two different sounds (two different instruments) in instrumental music. One solution is to introduce the third instrument whose quality falls in between those of two concerned instruments. This approach can be found in the third piece of Webern's *The Five Pieces for Orchestra*, Op. 10. Erickson analyses the first and the last four bars of the piece and states that the sounds of bells, less pitch specific than the other instruments, which are used as mediating elements, bridging pitched instruments (mandolin, guitar, celeste, harp, harmonium) with percussion that has a high degree of noise (bass drum, snare drum).

Figure 3-4-2-3: The beginning of the third piece of Webern's *The Five Pieces for Orchestra, Op. 10*

Another technique applied in this piece that enables a smooth interpolation between the sounds found within it, is the way in which the edges of the different instruments have been obscured by reducing the qualities of the attacks. “The dynamic marking of PPP certainly enhances the mixing of the sounds by minimizing the effects of the attack transients—they tend to mask one another”. (Erickson, p. 166)

I have already mentioned in chapter 1 how the third piece of Schoenberg’s *The Five Pieces for Orchestra, Op. 16* also exhibits the same sort of reduced attack qualities. Regarding the piece, the composer’s note to the conductor says, “The change of chords in this piece has to be executed with the greatest subtlety, avoiding accentuation of entering instruments, so that only the difference in color becomes noticeable.” (Erickson, p. 37)

Another technique used in instrumental music to smooth transition from one instrument to the other is to make the phrasing (of pitch) and the change of instruments asynchronous. I have mentioned in the previous section that Berlioz's *Symphonie Fantastique* and Bach-Webern’s *Ricercare* employ this method.

The electronic music made it possible to interpolate sounds with much higher resolution. In Stockhausen’s *Gesang Der Jünglinge,* the seamless interpolation from the real voice to electronic sounds is intended. Stockhausen divided the four basic materials (voice, sine wave, noise, and impulse) into 12 categories.

1. Sine tone complexes
2. Impulse complexes
3. Sounds and syllables
4. Noise filtered to about 2% wide (in Hertz)
5. Single impulses
6. Synthetic vowel sounds
7. Noise filtered 1-6 octaves wide
8. Showers of impulses of statistically fixed density, filtered 1-6 octaves wide
9. Single impulses in chords
10. Chords from 2% (in Hertz) wide noise bands
11. Sine tone chords
12. Sung chords (Stockhausen,1964, p. 59-60)

This categorization can be interpreted as different stages of interpolation. Smooth interpolation between voice and electronic sounds should not be easy considering the limited equipments available at the time. However, Stockhausen made this level of interpolation possible by analyzing phonetic qualities of the voice, finding similarities between different types of voices (e.g. vowels and consonants) and different types of electronic sound (e.g. sine wave and noise). Stockhausen then proceeded to establish various mediating states using this elaborate compositional scheme.

As mentioned in chapter 1-5, Koenig, who had helped Stockhausen to realise *Gesang Der Jünglinge*, classified electronic sounds into different categories (e.g. harmonic sound, noise, impulse) and summarized transformation techniques between them. The fact that such transformations are available gave him a vision of ”unbroken continuum of all timbres”.

Today, various signal-processing methods allow for sophisticated morphing between different qualities of sounds as well as this, infinitely fine resolution of interpolation (in sample level) is now available. Trevor Wishart, is a composer who exemplifies this by making many pieces that are primarily concerned with sound transformations happening in conjunction with the development of new signal processing methods.

The convolution of Band-Limited Oscillator and other sounds explained in the chapter 2 is my own approach to make continuous transformation between different sonic qualities with controllable interim states. I would like to add that changing envelope could also be considered as the interpolation of timbre, because the shape of envelope has strong influence on timbre perception.

While many of today's software and technologies have facilitated the creation of sound metamorphosis — to happen on a very high level of acoustic resolution — comparable compositional systems, that can realize sophisticated timbral interpolation, have been far more difficult to achieve. Given this it seems that if we could invent a technique that produces convincing sequences of different timbre, such a system may be labeled as a "compositionally" sophisticated system of timbre interpolation. In this case, the two sound materials to be interpolated would be analogous to lower and upper limits of a pitch sequence. The change of mediating timbral states could then correspond to a change of pitch and indeed, attempts to make sequences of timbre (that have been sought for years), could finally be realized.

Historically, it seems that it was Schoenberg who came up with this concept, which is evident in his Klangfarbenmelodie — this concept clearly demonstrates his desire to treat timbre in a comparable way to the treatment of pitch. Schoenberg states:

> If it is possible to make compositional structures from timbres which differ according to pitch, structures which we call melodies, sequences producing an effect similar to thought, then it must be also possible to create such sequences from the timbres of the other dimension from what we normally and simply call timbre. (Erickson, 1975, p. 105)

Webern, a student of Schoenberg, seems to have worked even harder than his teacher on realizing this goal. The orchestration of Bach's *Ricercare* is one good example of sequencing instruments of different timbres in a systematic way. It deserves mentioning that Koenig was also very much impressed by the schematic approach Webern employed in this orchestration. The impact of Webern's work is clearly felt in *The second phase of electronic music*, as Koenig explains how Webern's scheme connected with his own interest in dealing with timbre as parameter.

> This detaching of individual characteristics from the totality of the acoustical impression is called the division into parameters. Timbre is one of them, in instrumental music at least. Others are, for example, pitch, duration or volume. In serial music, these parameters have been subjected to the most exact control, that is to a scheme not unlike Webern's, as just demonstrated. (Koenig, 1965b, p. 3)

Wessel's concept of timbre-space also aimed at establishing techniques for sequencing timbre. His basic idea can be illustrated in the cited figure seen below.

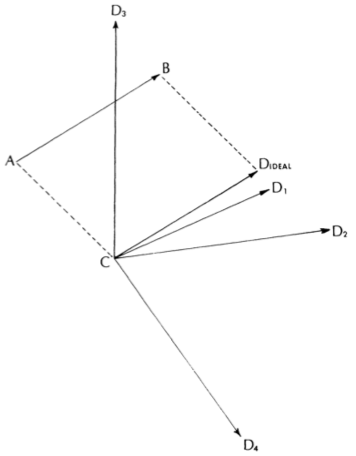


Figure 3-4-2-4: Parallelogram model of timbre analogies. A —> B is a given change in timbre; C —> D is a desired timbral analogy, with C given. (Wessel, 1979)

This figure (3-4-2-4) is a parallelogram of a scheme to make the transposition of a sequence of possible timbres. The parallelogram represents the way in which other sequences of timbre can be geometrically parallel to one another, within a related timbral space.

Although the sequencing of timbre was dreamed and attempted by some composers, a solid theory or widely adopted method for this manner of composing has not been created up until the present. Despite Schoenberg's attempt in op.16, he never personally thought that he successfully composed a Klangfarbenmelodie. (Cramer, 2002)

Stockhausen is another composer who was very preoccupied by the parameter of timbre, in particular he used a various schemes related to timbre that suited him for a given composition. Koenig also applied his own serial-ideas onto timbre when he added sine waves to make spectra, a process he referred to as being the first phase of electronic music. However, both of these composers did not make a generic scheme for sequencing timbre, namely, a system of timbral control that can be to used in various compositions. It is also hard to say that Wessel's concept of timbre-space is a commonly used technique in composition.

Albeit I am not proposing a grand theory of how to sequence the parameter of timbre, instead, I would like to suggest a possibility to employ timbral movement within algorithms, and as I have mentioned in the previous section, this will be arrived at in order to achieve sophisticated-levels of interpolation — where boundaries between two extreme sonic states can be integrated. This means that regardless of signal processing techniques used, interpolation will be able to be done by changing parameter value(s).

In this way, dealing with timbre as a parameter will suggest that there can be a determined range between two extreme states, which is in essence, a idea that is rooted in serial-thinking. Thus depending on the horizontal division, interpolation can be done either continuously, irregularly, or in a stepwise fashion. That being that in the case when stepwise timbral interpolation is chosen a time grid can consequently determine the relationship between timbral change and rhythm. Vertical division will then determine the resolution of mediating states between these two extremes and movements that determine this type of gesture will also help to establish a process of such interpolation. In summary of this thought, it should be mentioned that applying algorithms, for the purpose of sequencing timbre, will be explored in more detail in my future work.

2. Adding common sonic quality and timbre movement

Every processing method produces its own characteristically sonic results. By applying the same processing methods to different sound materials, we can imprint a commonality in resultant sounds. We can also implement common timbral movements into different materials by applying a similar type of processing.

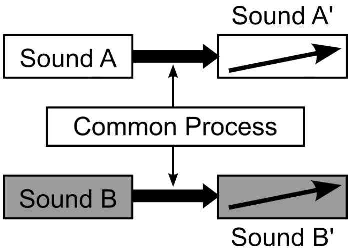


Figure 3-4-2-5: Adding common sonic quality

In figure 3-4-2-5, a common processing method is applied to different materials Sound A and Sound B. The resultant Sound A' and Sound B' bare the earmarks of that processing method, and they could share similar timbral movement. For example, applying band pass filters to different materials and narrowing the bandwidth creates a common timbre movement. Applying the same type of reverb to different materials is another example of adding a common sonic quality. This implies that room acoustics can be used as an adhesive agent of multiple materials. If we increase the output levels of both materials to reverb at the same time, a common timbral movement occurs. Moreover, considering the fact that the amplitude envelope has strong influence on timbre perception, applying the same low frequency amplitude-modulation to different materials is also an efficient way of giving them a common quality. This can be done by using a low frequency oscillator, low frequency noise, or the envelope follower that is applied to an external material.

This way of connecting different materials is my approach to the issue that people tend to listen to technology in the music. Listeners who know synthesis techniques have a tendency to detect synthesis methods used in electronic music composition instead of purely enjoying the music. This is the flip side of situation that sounds made with a particular synthesis or processing method have common characteristics. It is inevitable that every signal processing produces distinguishable results will subsequently lead listeners to notice what kind of processes have been used. I believe what composers can do is not to hide the hallmarks of such processing methods, but rather should utilize them in a creative way. My own creative approach is to make use of the peculiar sounding results of certain signal processing methods, as a way of revealing the different materials inherent in the tools so that the technology can be "heard".

I have explained about adding common timbral movement so far. Extending this idea, it is possible to make same kind of "timbral counterpoint". If one material from a movement of timbre follows that of another material, for example by changing the bandwidth of filters applied to each material with some time interval, then this means that it can also be regarded as a type of timbral canon. It is also possible to implement contrary timbre movement in each material. For example, we can apply a filter with widening bandwidths to one material and a filter with narrowing bandwidths to another material. This can be seen as being analogous to melodic inversion of counterpoint.

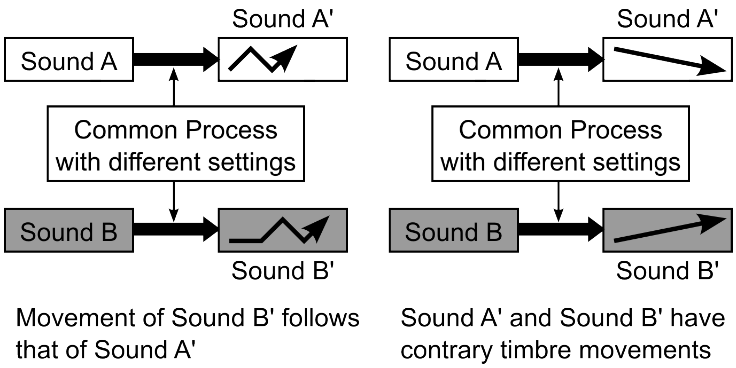


Figure 3-4-2-6: Timbral counterpoint example

3. Introduce a level of interaction between different materials

The third way of making timbral connection between materials is to introduce some action and reaction.

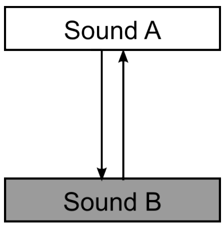


Figure 3-4-2-7: Interaction between materials

To make interaction between materials, I have developed a technique called "timbral side-chaining". It makes timbral movement of one material react to a timbral gesture of another material. Side-chaining is a technique that takes one sound as a control source and uses it to manipulate another sound. It has been used when making voice-overs for *ducking*s, a process where the volume of music is lowered when dialogue occurs. This technique is also popular in dance music production. One of the typical situations applying side-chaining in music production is to use the amplitude of kick drum sounds as control source and compress the dynamics of bass sounds when kick comes in to avoid the excessive low frequency. We can consider this as a method to establish amplitude relations between two different sound materials. Side-chaining is also used to change the cut-off frequency of filter applied to one sound reacting to the amplitude envelope of another sound. This case can be considered as making relations between a sound's amplitude and another sound's timbre.

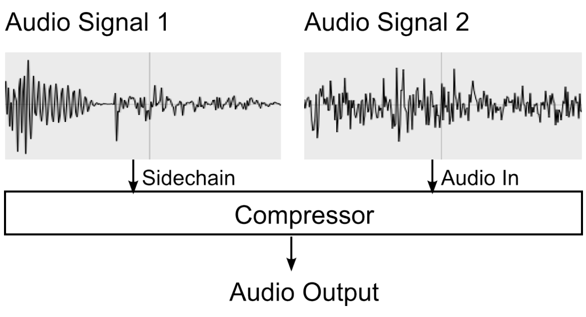


Figure 3-4-2-8: Sidechain compression—compress the audio signal 2 based on the level of audio signal 1

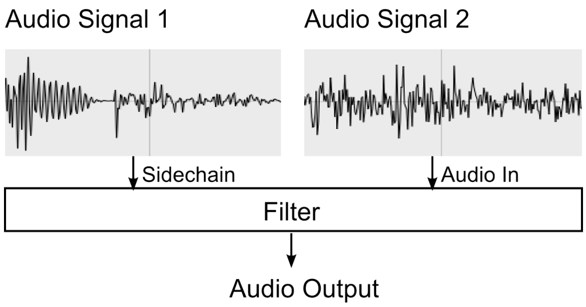


Figure 3-4-2-9: Sidechain filtering—cutoff frequency of filter is controlled by the level of audio signal 1. Audio signal 2 passes through this filter and frequency component is changed accordingly.

We can apply similar techniques to control the parameters of the Band-Limited Oscillator. The change of timbre is achieved by modulating the parameters of the Band-Limited Oscillator (*loHarmonics, hiHarmonics, slope, evenOddRatio*) in reaction to the amplitude fluctuation of another audio signal.

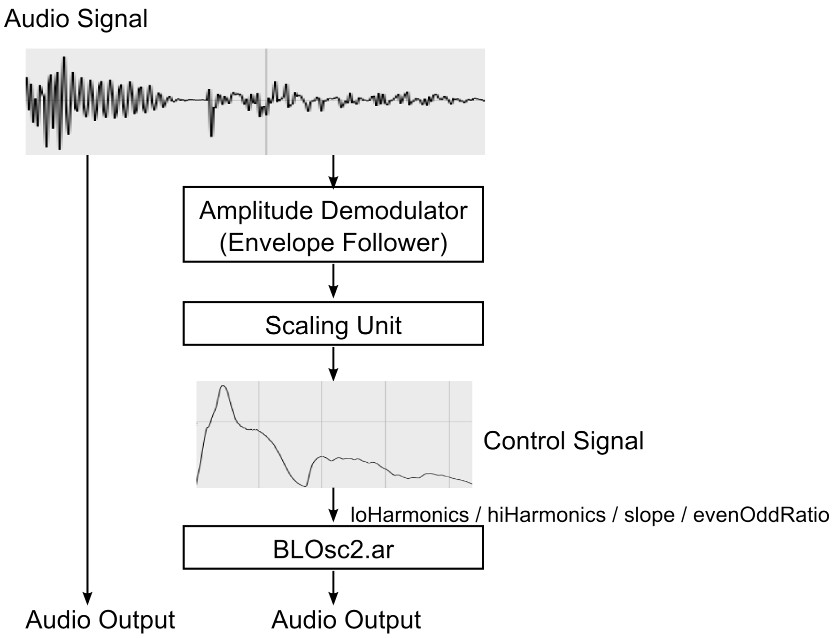


Figure 3-4-2-10: Timbre side-chaining with Band-Limited Oscillator—Control signal based on the level of an audio signal modulates parameter(s) of BLOsc2.ar.

In SuperCollider, the combination of Amplitude.kr and Lag.kr can be used as an amplitude demodulator unit. Lag2.kr or Lag3.kr can also be used instead of Lag.kr.

I have shown the methods of making "amplitude relation between two different sound materials" and "relation between one sound's amplitude and another sound's timbre". It is also possible to establish "timbral relation between two different sounds". For example, the following patch works for this purpose.

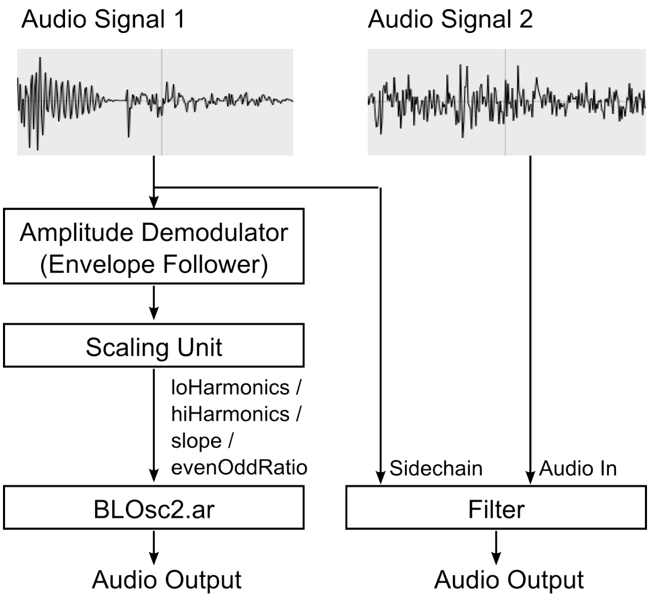


Figure 3-4-2-11: Example of relating timbre of two sound—The level of audio signal 1 controls both the parameter(s) of BLOsc2 and the cutoff frequency of filter applied to audio signal 2. In this case, the audio outputs from BLOsc2.ar and the audio output from filter have a timbral relationship.

However, I have come to acknowledge that side-chaining is not only effective for dance music, but also has a large potential to be explored more extensively in the composition of electroacoustic music; able to create organic and complex structures whose parts can act and react to one another. This technique is used in my composition *Colour Composition 2 and 3*, and this detail will be explained in the next chapter.

3-4-3 Sound transformation and consistency

At present I have so far explained the methods of giving consistency to my work by relating different materials. But within this overall concept, my aim is to find ways to make timbre to be a more active musical parameter. Thinking about this situation from a different perspective, perhaps we can devise rules for sound transformations to obtain diverse materials while still maintaining an overall compositional consistency. The important role of transformation in composition is to make a number of materials that share common characteristics. As it was mentioned earlier, in chapter 1-6, Koenig used systematic transformation to make variants that have neighbouring relationships. Koenig states, “During sound transformation common characteristics are caused by the formation of variants because of similar operations in the existing structure.” (Koenig, 1965a, p. 11)

I would like to suggest two approaches for transformation that clarify the common characteristics between materials.

(1) One material, various processing methods
(2) Various materials, one processing method

In approach (1), the usage of various processing methods enables a composer to obtain diverse sounds. While at the same time, all the derived sounds have coherent characteristics because the source material is same. This is the approach Koenig used in his compositions. In my opinion, this method is related to the interpolation of timbre as it has been explained in previous sections. Fixing source material corresponds to first determining the mediating state of interpolation. This is because obtaining derivatives with series of processing essentially corresponds to moving extreme states.

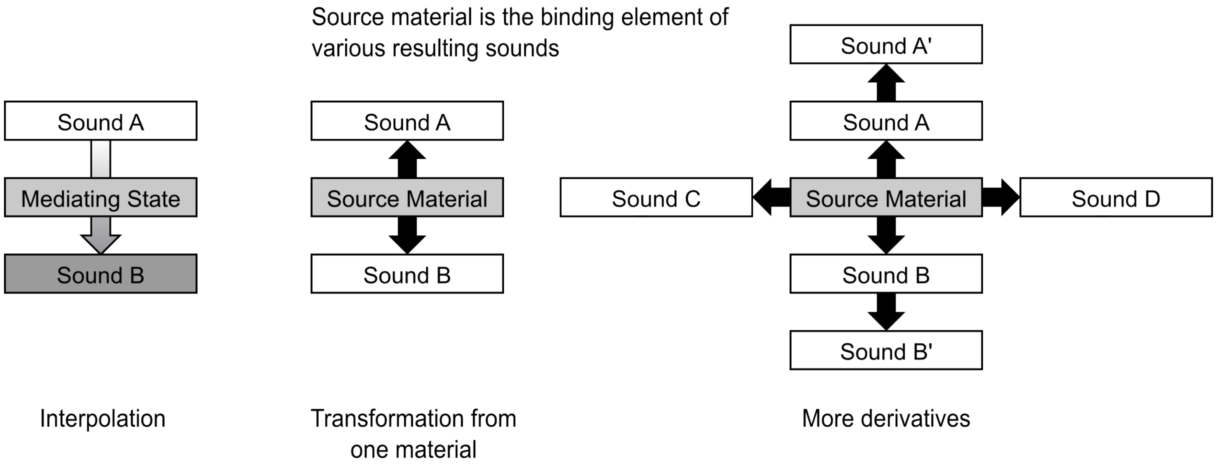


Figure 3-4-3-1: One material, various processing methods

In approach (2), the diversity of resultant sounds comes from the fact that various source materials are used. The consistency comes from the fixed processing method. One example is to apply chopping or granulation to many field recording materials. This approach corresponds to adding common sonic qualities to different materials that I have explained in the previous section.

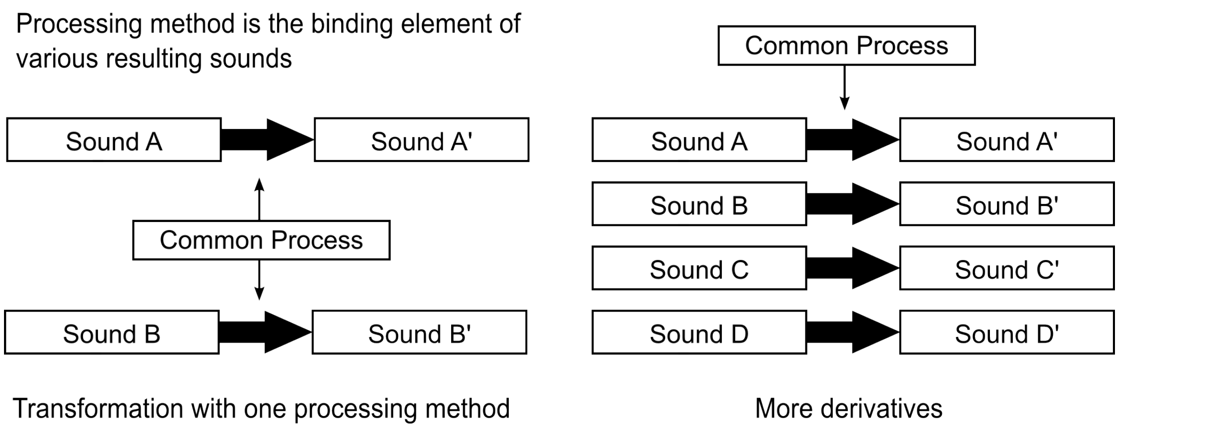


Figure 3-4-3-2: Various materials, one processing method

In my piece *Flock*, I have tried both approaches. In some section, I processed the recording of a fluttering bird with various methods, such as transposition, filtering, and phase vocoding. In another section, I used various materials for one processing method, namely phase vocoding. This will be explained in more detail in chapter 4-1.

One important aesthetic judgment required for composers who work with transformation is to find an acceptable limit of processing. Koenig emphasizes the importance of maintaining audible links between the source material and resulting sounds after the series of transformations. Koenig says in relation to his composition, *Terminus*, “The basic sound's characteristics were not allowed to be neglected.”(1971, p.7) Suppose we are taking the first approach I have explained: *one material, various processing methods*. During the course of repeating transformations, the characteristics of basic material is gradually disappearing. Too many transformations or too radical a process of transformation results in completely erasing the connection between source material and generated material. This situation does not make any differences from introducing new material and it impairs the consistency of composition. This was also emphasized in a personal communication I had with K.Tazelaar (September 23, 2014) as he suggested that the lowest end of the generation diagram of *Terminus* is where the edge between links of source sound and generated sound become barely audible.

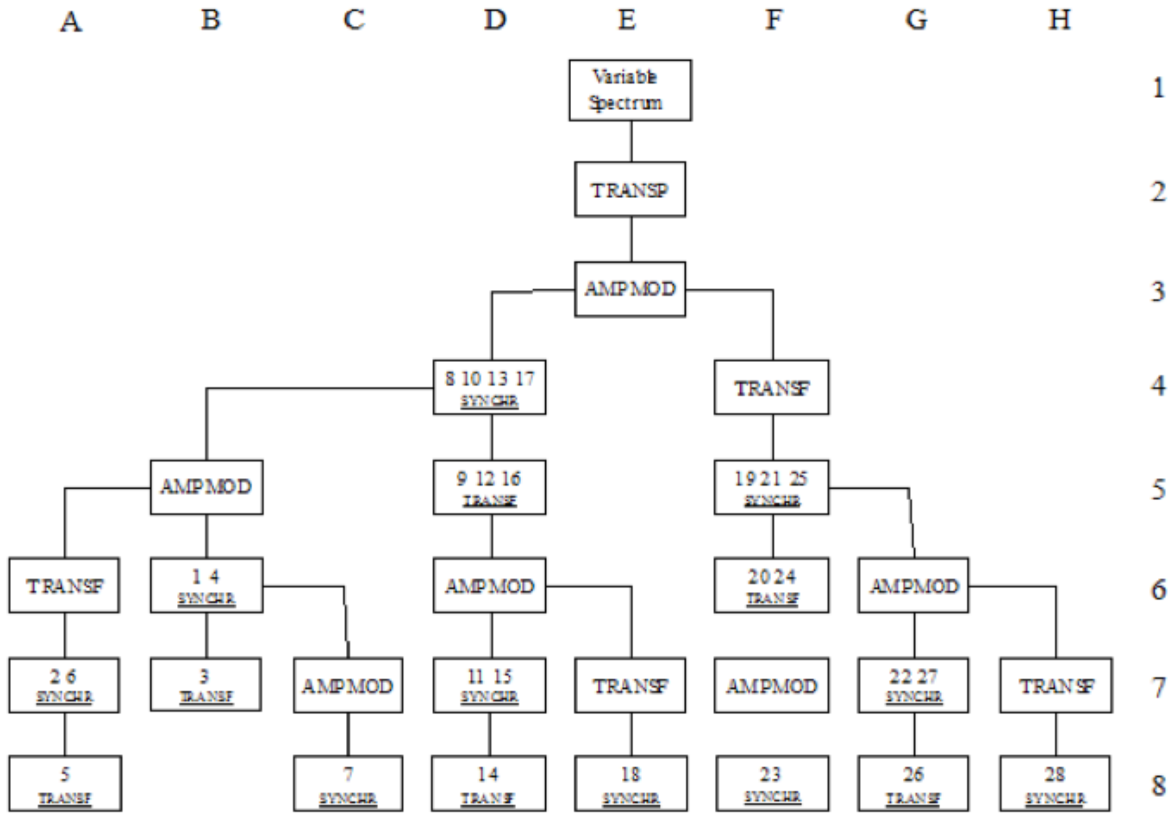


Figure 3-4-3-3: The generation diagram of Koenig's *Terminus* (Koenig, 1971, p.8)

Trevor Wishart is a composer of a more recent generation who has also discussed the importance of establishing audible relations, during the course of transformations, between a source sound and goal sound. In this respect it is interesting to see how Wishart uses many new signal processing techniques that Koenig did not use.

> The infinite malleability of sound materials raises another significant musical issue, discussed by Alain Savouret at the International Computer Music Conference at IRCAM in 1985. As we can do anything to a signal, we must decide *by listening* whether the source sound and the goal sound of a compositional process are in fact at all perceptually related, or at least whether we can define a *perceptible* route from one to the other through a sequence of intermediate sounds. (Wishart, 1994, p. 23)

Wishart introduced Savouret's another idea distinguishing *source-focused transformation* (where the nature of the resulting sound is strongly related to an input sound) as well as *process-focused transformations* (where the nature of the resulting sound is strongly determined by the transformation process). This categorization is comparable to two approaches of transformation I have previously explained. Wishart then claims, "In general, process-focused transformations need to be used sparingly . . . process-focused transformation can rapidly become clichés".(Wishart, 1994, p. 24)

However, I have a different opinion about this issue given that I see that sound can be made by a particular process, and even if this process is perceived as cliché, the problem is not the process itself, but essentially the composition. The same applies for synthesis. For example, if the sound of granular synthesis is perceived as cliché, what is actually a cliché is not the granular synthesis itself, but the way a composer uses such a technique. It does not make sense to say that the sound of particular instrument is cliché, nor does it make sense to say that the sound of a particular synthesis or processing is a cliché. Regardless of acoustic instruments, synthesis, and processing, there are sounds that can be achieved only by particular method. In other words, there are unique gestures or forms that can be obtained by particular way. In my own opinion, I feel that clichés can be averted if a composer find the valid reason to use the gesture made with particular process in the unique context.

3-5 Increase complexity by way of timbral movement

Systems bind different materials and by this act it can be said that the act of binding materials is not only effective for making consistency but also for increasing overall complexity of music. From an aesthetic position, I often regard music having enough complexity as being interesting, however, this raises the broader issue of defining exactly what complexity is? Koenig discusses his ideas about complexity in the article *Complex Sound* and based on the ideas written in this article, I would like to organize my own interpretation surrounding the idea of complexity. Furthermore, in doing this I hope to imply that complexity can be divided into the following two principle categories within electronic music, that of *Sonic Complexity* (Complexity in one material) and the other representing *Formal Complexity* (specifically the type of complexity caused by the aggregation of multiple materials).

Within my work I am interpreting "sonic complexity" as the complexity of one material, and "formal complexity" as the complexity caused by the aggregation of multiple materials. Although this division is based on Koenig's idea, his viewpoint emphasizes time scale to a larger degree. This is apparent in the way that he categorizes "microtome" as a description of the sound and "macro time" being a generalized description of form. According to Koenig, formal complexity had been concerned in instrumental music and early electronic music. On the other hand, sonic complexity became an interest only after the development of equipment. Koenig states, "In the first two phases of electronic music, complexity could only be controlled in macrotime; automation and programming also permit control of complexity in microtime" (Koenig, 1965a, p. 2). The first phase of electronic music is when synthetic sounds were made only by adding together sine tones. However, in the second phase coinciding with the standardization of studios, various waveform generators and other effects became available. The third phase is, according to Koenig, chiefly characterized by the automation of making electronic music as well as the emergence of programming of sound using computers.[1] In light of this history, I would like to examine the two categories of complexity, which I proposed (sonic and formal complexity) in greater detail.

3-5-1 Sonic Complexity

In chapter 1-6, Koenig's opinion is introduced to illustrate that even individual sounds can be regarded as objects formed in the electronic music. From this viewpoint, we can deduce

1 In *Complex sound (1965a)*, automation is categorized in the third phase. In *The second phase of electronic music (1965b)*, Koenig categorizes automation in the second phase.

that even individual sounds have an inherent amount of complexity. This is “sonic complexity”.

According to Koenig, sine tone represents a minimum of complexity because

(a) the pattern is the simplest imaginable,
(b) the rule for succeeding patterns is also the simplest: all succeeding patterns are congruent to the first (Koenig, 1965a, p. 2).

Regarding sonic complexity, I consider Koenig’s two criteria to be useful in regard to formulating a definition of what principally constitutes timbre. If we consider (a) pattern in time domain, it is waveform. If we see it in frequency domain, it is spectrum. (b) the succession of patterns is rephrased as the evolution of waveform in time. Thus, these criteria correspond to two physical aspects of timbre explained in chapter 1. The evolution of sound, or the change of waveform, is the important aspect of sonic complexity. In my opinion, both the microtime and the macrotime evolutions of sound (timbre) contribute to the sonic complexity. Using the Band-Limited Oscillator I developed ways to enable the evolution of timbre by moving parameters, and this parameter movement directly produces a certain amount of sonic complexity.

Koenig also used the term “constant complexity” and “variable complexity” referring to *Klangfiguren*. In *Klangfiguren*, constant complexity was created when making basic materials. The methods of making basic materials for *Klangfiguren* include the construction of spectrum and the change of speed. Spectrum was made of sine waves whose frequencies were arrived at through various serial processes. On the other hand, variable complexity was created by transformation. Transformation methods in this piece include ring modulation, group formation, the insertion of pauses, transposition, superposition, reverberation, and the application of a variety of enveloping techniques. (Koenig, 1971)

Although the categorization of “constant complexity” and “variable complexity” is not completely equal to two criteria of sonic complexity, (a) basic waveform and (b) evolution of waveform, it seems to be they came from the same idea. This categorization may also be related to my categorization of “*short term, non-directional timbre movement* ” and “*long term, directional timbre movement*” because both categorizations concern themselves with whether the change of sound has contributed to the sonic evolution or not.

Taking account of the fact that sonic complexity became the interest of composers only after the development of automation and computer, the following idea sounds like a contradiction; we can learn techniques to enhance the sonic complexity of electronic music from instrumental music. Surely, microtome complexity is difficult to control in instrumental music. This can only be done by players to give “nuance” to music and composers do not have free rein over it. However, it does not mean instrumental music does not have sonic complexity. In fact, instrumental sounds have very complex waveforms and is full of microtome timbral changes. If we could obtain both the innate complexity of instrumental sounds and the controllability over it with the help of computer programming, we could achieve rich sound structure. From here, I would like to examine why the sound of instrumental music is rich and if there are ways to obtain this richness in electronic music using currently available techniques.

Why is instrumental sound rich?

As explained in chapter 1-7, the microtime spectral evolutions contribute to the attractiveness of instrumental sounds. Today’s technology enables simulating this characteristic in electronic sound to certain extent. For example, we can use additive synthesis and change the amplitude envelope of each partial. Physical modeling is another way of simulating microtime spectral changes. Of course, the efficient control over partial amplitudes in additive synthesis is a difficult issue (Band-Limited Oscillator is one approach to solve this issue) and not all sounds are easily simulated with physical modeling. Technical means to synthesize sounds with microtime spectral changes still need to be developed further. Under present circumstances, I am using Band-Limited Oscillator and some physical modeling UGens in SuperCollider (such as Klank) to make time-variant spectrum, however, their sound making capacities still have limitations.

Another important element giving richness to instrumental sound is inharmonic contents. Normal oscillator sounds (including Band-Limited Oscillator) do not have inharmonic partials, ones that natural sounds and instrumental sounds contain. Adding slightly detuned oscillator sounds in subtractive synthesis, ring modulation, and frequency modulation with non-integer harmonicity ratio can be considered as techniques to give richness to electronic sound by adding inharmonic contents.

I need to make a supplementary statement regarding the above, specific to the notion that the oscillator's inherent sonic characteristic does not imply that they are characteristically undesirable, even though they do not produce any spectral change over time nor inharmonic content. In actual fact, the clean sound quality, which cannot be obtained with other methods of sound production, offers to composers a uniqueness of sonic character that can be constructive in a variety of ways.

Why is the orchestral sound rich?

A sound of one acoustic instrument already contains microtime evolution. Ensemble sound composed of many instruments has even more complexity. Considering string section of an orchestra, every player makes a sound with slightly different timing and intonation even though they play a tutti part. In addition, players are spread across the stage, making a spatial effect. The amalgamation of multiple sounds with slightly different timing, articulation, and spatialization provides microtime sonic fluctuations and it contributes to the richness of orchestra sound.

Moreover, certain composers have proactively made use of the ensemble of strings with different movements and created rich sound mass. Xenakis's *Pithoprakta* and Ligeti's *Atmospheres* are good examples. In those pieces, each string player is indicated different movement. Accordingly, intentional deviations of sonic components constituting a bigger sound mass are created. This mechanism can be simulated in electronic music by adding multiple of oscillator sounds with slightly different parameter settings. This idea is applied to the opening of my piece *Colour Composition 2* to make rich sound. The details of its technique will be explained in chapter 4-2. Instead of adding sounds made of same synthesis method but with different parameter settings, it might be also interesting to play multiple sounds made of different synthesis or processing methods with similar parameter (e.g. pitch and rhythm) settings. I am planning to try this idea in my future work.

3-5-2 Formal Complexity

On the other hand, I regard the formal complexity is derived from the following criteria.
(a) Number of sound materials (layers)
(b) Relationships between sound materials (layers)

Relationship is the important keyword to understanding complexity. Koenig discusses such relationships and how they occur at different levels.

To paraphrase Koenig, it can be seen that in connection to this, it is also necessary to speak of higher complexity especially if the individual parametrical values have many relationships with one another, and are of a low complexity so that only a few relationships are clearly identifiable.

Thus complexity can be observed as follows:

(1) in a single parameter,
(2) (a) between two (or more) parameters of a form-section,
    (b) between all parameters of a form-section,
(3) between several form-sections. (Koenig, 1965a, p. 6)

In Koenig's viewpoint, relationships can be produced within a parameter of single sound material by comparing the alterations in the given parameter. Relationships can also be made between multiple parameters of one sound. However, I would like to focus mainly on relationships between different sound materials. From a serial viewpoint, the relationships between sound materials are examined and perceived by comparing parameters of both materials. For example, in instrumental music, passages played by different instruments are sound materials, which are connected through rhythm, counterpoint, chord progression, etc. This means, materials are connected based on systems of time, pitch, harmony, and so on. However, there is a severe limitation to connect materials through timbre though some composers such as Webern have attempted.

In the electronic music, however, different materials can create mutual relationships through timbre at the moment they are generated and transformed.

For example, I would like to consider voltage controlled patches as a method to create timbral relationships when generating materials. A relationship between different modules is created when the same control signal is applied to parameters of both modules. If this is to happen a control signal will modulate frequencies of multiple oscillators resulting in pitch relationships established among these modules. Moreover, if this occurs sound control signals will modulate cutoff-frequencies of other multiple-filters, an affect of this will be that timbral relationships will be created. This scheme is analogous to one of the systems I discussed in the previous section that applied similar processes to different materials. In voltage controlled patches, organic musical behavior can be created by constructing networks of related modules. Figure 3-5-2 is the example of voltage controlled patch K. Tazelaar showed in the Institute of Sonology. (May 14, 2013) Random control voltage from the left tendency mask generator is applied to both the voltage quantizer of the left groups of modules and the sample-and-hold of the right groups of modules. Voltage from the right tendency mask generator goes the other way around. This cascade network of modules results in interesting relationships between two audio outputs, thus it contributes to the complexity.

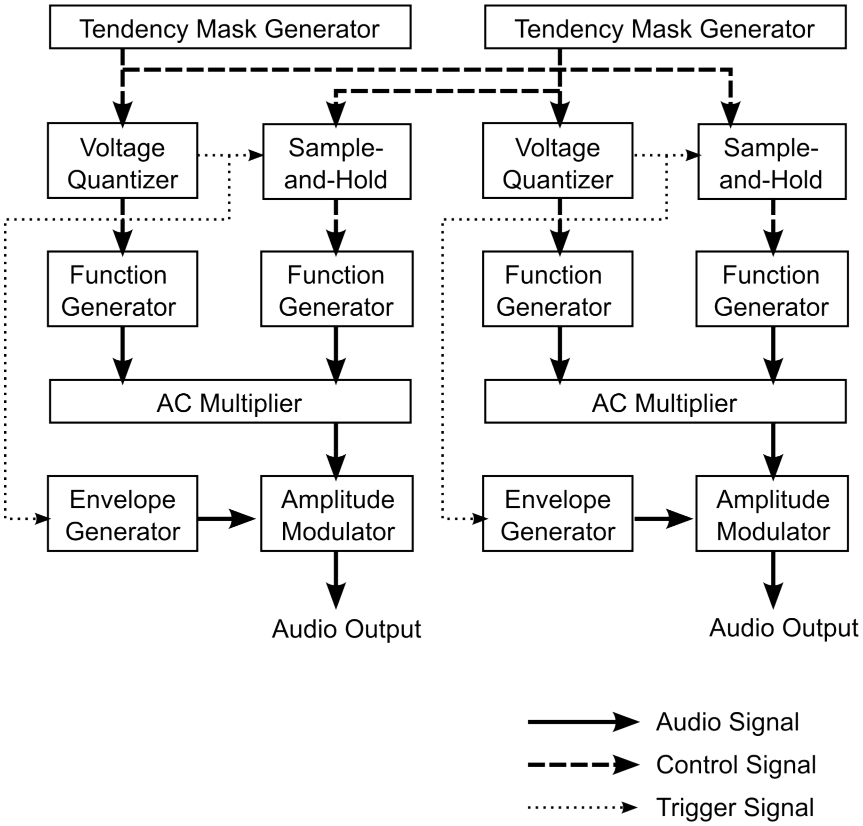


Figure 3-5-2: Example of voltage controlled patch exhibiting networks of modules

Proceeding with this discussion further, the topic of complexity being arrived at through transformation needs to now be addressed. As I have explained in the previous section the important role of transformation is to make variants share common characteristics. As mentioned in chapter 1-6, Koenig regarded sound transformation as a method to establish vertical as well as horizontal neighbouring relationships. Koenig's work, *Terminus,* exemplifies the networks of such materials created by succeeding generations of these type of transformations. This approach can be considered as the act of increasing formal complexity (complexity in macro time).

Sound transformation can also be used to increase sonic complexity and in connection to this Koenig states the following: "By means of transformation of the material, the elementary timbres become complex" (Koenig, 1971, p. 3) referring to compositional methods he used in *Essay*. It seems he was concerning sonic complexity (complexity in microtime) in this sentence. In this piece, elementary timbre includes sinus tones, noise, impulse, as well as the combination of all of these elements. Transformation methods include ring-modulation, transposition, filtering, reverberation, enveloping, distortion by overloading, and tape-splicing. In *Klangfiguren*, as I have already mentioned, "variable complexity" is achieved also by transformation.

At this point, it is been thoroughly demonstrated how sound transformations can play a variety of roles within electronic music. This can include obtaining levels of sonic diversity, achieving formal unity, enhancing and adding to perceived sonic and formal complexity, as well as giving a composition a sense and an actual degree of musical evolution. To sum up the discussion so far, I would argue that the following keywords, originated by Koenig, are not discrete concepts but closely related to each other. These would include concepts such as "composing sound", "fluctuating timbre", "form", "complexity", and "transformation". "Composing sound" deals with not only the creation of waveform or spectrum, but also the creation of sound that evolves over time. The evolution of sound is related to the concept of "fluctuating timbre". Timbre fluctuation articulates the material's "form" and creates "sonic complexity". At the same time "formal complexity" is created when timbral relationships are implemented among multiple materials — whereas "transformation" can be seen as the act of making both "sonic complexity" as well as "formal complexity".

In the closing of this section, I would like to write why I have obtained my interest in establishing relationships between materials. I think it came from my background as a DJ. What a DJ should do is to find the relationships between different tracks and make a meaningful flow of sound. From the analytical point of view, this is the act of finding proximity between musical elements used within different tracks. For example, two tracks have relationships if they use similar drum patterns, similar bass movements, sample the same instrument, use similar synthesizer sounds, or possess other related effects. DJs then make larger structural relationships by connecting tracks that have similar characteristics, often playing them in sequence and sometimes stacking them vertically. In the previous section, I introduced my compositional approaches to connect materials by interpolating them, adding common sonic qualities to them by introducing interactions. For me, selecting tracks as a DJ and processing these materials as a composer can be viewed as a contiguous activity, especially in terms of making networks of materials that are connected to each other through similar sonic qualities or related by movements that are able to achieve macro scale unity and an overall musical complexity.

3-6 Timbre movement and other parameters

In relation to enhancing sonic complexity, it is also necessary to discuss how the movements of other parameters, such as pitch and rhythm, often make listeners less sensitive to movements of timbre. With this in mind, Koenig 's argument about how our hearing is more sensitive to moving parameters than fixed parameters becomes particularly relevant.

> If several parameter fields are superposed to form a structure, dominating characteristics can occur which encumber or prevent the perception of other parameters. It is therefore sensible to arrange the parameters concerned according to their hierarchy, so that any "dominants" (main parameters) allot an unambiguous function to the single structure in a sequence of several structures.
> As a rule, fixed parameters recede from other parameters which change frequently. (Koenig, 1965a, p. 5)

From this perspective, Koenig expresses clearly the importance of composers to effectively judge which parameter should have a dominant role and upon doing this the emphasized parameter also need to have clear movements allotted to it, in order to prevent an overall perception of sonic ambiguity. In this case, in regard to my own work, timbre can be given a dominant role, suggesting that it should exhibit lucid movements that supersede pitch, rhythm or other parameters.

However, through my own compositional experiments it has been evidenced that clear movements of timbre can be perceptually much more subtle, or less obvious, than those movements of pitch or changes in rhythm. For example, if a composer simultaneously changes aspects of pitch and timbre, a listener's attention tends to focus on pitch movement. Given this tendency of musical perception, it is necessary to suppress pitch movements and changes in rhythm for the sake of highlighting variations, evolutions and overall changes within the parameter of timbre. Therefore, slowly changing music, or repetitive music, successfully uses this approach in order to shift a listener's attention from focusing on macro structure relationships to a more attuned sense of perception that is much more aware and sensitive to microtime timbral evolution.

Trevor Wishart, who has spoken about how this issue functions within his music and the work of others, confirms the validity of this strategy as a way of drawing more attention to the parameter of timbre: “A minute audible feature of a particular sound can be magnified by time-stretching or brought into focus by cyclic repetition (as in the work of Steve Reich).” (Wishart, 1994, p.13)

Slowly changing music

I would like to take a look at concrete examples, starting first with the third movement of Schoenberg's *Five Orchestra Pieces,* Opus 16, “Harmonic and melodic motion is much curtailed, in order to focus attention on timbral and textural elements”. (Erickson, 1975, p. 37) The slow tempo helps listeners to strain to hear subtle changes of texture.

Works by spectral composers, such as Gérard Grisey, also exhibit slow movements. Accordingly, changes of harmony in their music are perceptually magnified. Depending on listener’s viewpoint, these changes of harmony may be considered as the changes of timbre. However, we need to be aware that for Grisey, the most important matter is not timbre, but the perception of musical time. As Grisey states: “spectral music is not a question of sonic color. For me, spectral music has a temporal origin. It was necessary at a particular moment in our history to give form to the exploration of an extremely dilated time and to allow the finest degree of control for the transition from one sound to the next.” (Grisey, 2000, p. 1)

In a similar fashion, the genre of Drone also emphasizes changes of texture by way of minimizing changes that take place in the parameters of rhythm and pitch. My own compositions, *Flock* and *Colour Composition 2,* also exhibit slow changes in these same parameters, with the intention to make listeners feel the detailed evolution of timbre.

Repetitive music

Repetitive music, such as minimal music, is another approach to suppress pitch and rhythm movements and magnify details. Steve Reich says in his 1968 manifesto titled *Music as a Gradual Process* that, “I begin to perceive these minute details when I can sustain close attention and a gradual process invites my sustained attention.” (Reich, 2002, p. 36)

Reich’s idea of “gradual process” may also be thought of as “suppressing big changes” and furthermore compositions of his, such as *Music For 18 Musicians,* allow listeners to concentrate on continuous changes of sound colour — largely because of the gradual processes inherent in the piece and this is specifically achieved by the prevention of major changes of pitch and rhythm.

Dance music is another example of repetitive music. For example, techno tracks are made of looping rhythms, and pitch movements are often limited. It is not uncommon that some techno-tracks are even entirely absent of pitch, as a lack of this parameter focuses a listener’s attention onto other sonic qualities and materials, such as slight changes of texture, instead of active perception of the macro structure.

Partly suppressing pitch and rhythm movements

On the other hand, Webern’s pieces have plenty of pitch jumps and frequent changes of rhythms while movements of timbre are still remain integral aspects of his musical intention. For this reason, at first glance it seems that suppressing pitch and rhythm, for the sake of enhancing the perception of timbre, is not relevant to his music. However, examining sections where Webern applied frequent changes of instrumentation reveal that he fixed both the parameter of pitch and rhythm in order to clarify movements of timbre. This is illustrated in the final measure of the first piece of *Five Pieces for Chamber Orchestra*, Opus 10, he demonstrated Klangfarbenmelodie (Erickson, 1975). The instrumentation changes at every note, including the doubling of flute and trumpet at the second note, overall this promotes the perception of a continuously changing timbre — while pitch and note duration remain essentially predictable or fixed.

Figure 3-6-1: The final measure of the first piece of Webern's *Five Pieces for Chamber Orchestra*, Opus 10 (Erickson, 1975, p. 16)

In the orchestration of Bach's *Ricercare*, Webern changes instrumentation in the repeating 8 bar phrase. This approach is similar to that of repetitive music. Moreover, the approach to avoiding changes of pitch and rhythm changes appear to exist only within parts of the composition where timbral change is being emphasized. In Webern's case, he elected this to use this strategy — avoiding the slowing down the overall tempo. However, in Stockhausen's piece *Mantra*, different techniques are used to draw attention to timbre, for instance he adds degrees of noisiness to the piano sound by using ring modulation in conjunction with a sine wave oscillator. Specifically, this can be seen at measure 57, where the noisiness of the piano is gradually modified, by shifting the frequency of a sine wave, while the piano continues to repetitively play a single note tremolo.

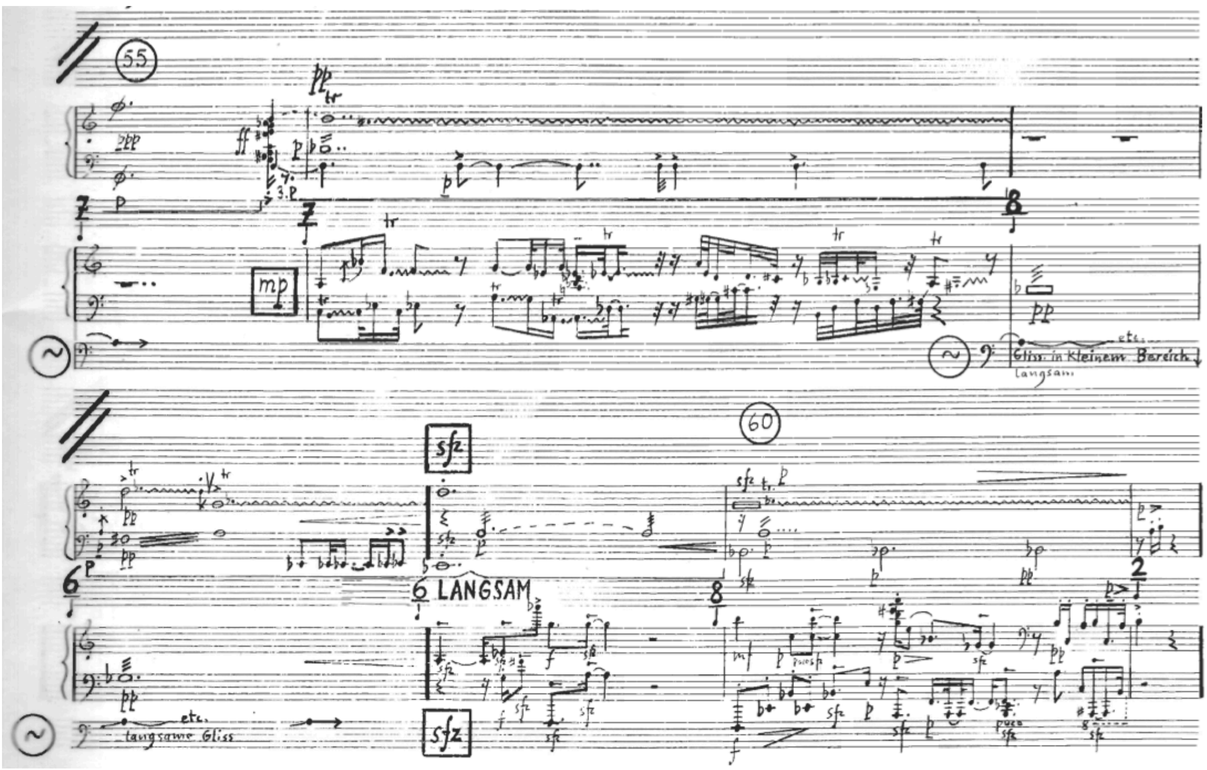


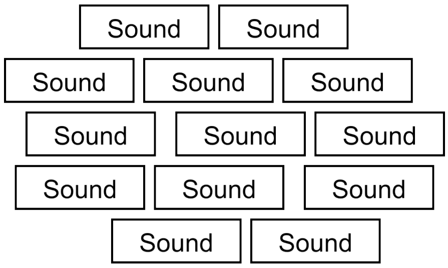
Figure 3-6-2: Excerpt from Stockhausen's *Mantra*

So far, I have discussed that listener's attention is not directed to timbre change when pitch and rhythm actively move. However, it does not mean timbral movement is useless in such moments. Fluctuating timbre still contributes to the sonic richness and the complexity even though listeners do not necessarily realize its presence. In this case, the role of timbral movement is sort of like the secret ingredient of sound, one that listeners may perceive at an unconscious level. Thus, it is required for composers to define the purpose of timbre movement at every moment of his or her composition in order to determine to what extent pitch and rhythm movements should be curtailed.

3-7 Layers, foreground, background

I have mentioned the following methods to increase complexity.
(1) Implement timbre fluctuation in materials (increase sonic complexity)
(2) Increase the number of materials, or layers (increase formal complexity)
(3) Establish relationships between materials (increase formal complexity)

I mainly adopt methods (1) and (3) for my own compositions. Method (2) was not widely used as it seemed to be a simple approach and one I have some difficulty with. This is because the more layers, the more difficult it becomes for listeners to pay attention to the minute features of each material. Even though only two layers are overlapped, if both of these layers possess a rich sonic complexity, it might be already difficult for listeners to perceive all the details of both materials. In addition, increasing the number of overlapping layers may impair the aural perception of being able to spot and recognize links between these various materials.



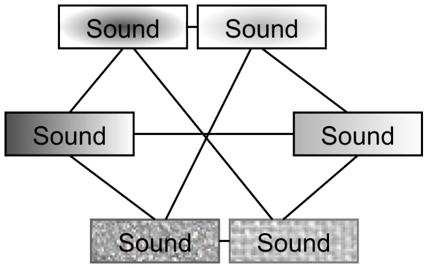


Figure 3-7-1: The image of complexity achieved by increasing the number of materials (left side of the figuree)
The image of complexity achieved by rich movements in materials and the relation networks between materials (right side of the figure)

In this section, I would like to focus on the superposition of materials this is because I believe it is necessary for composers to carefully highlight the foreground and background in order to effectively overlap layers. In principle, material in the foreground will attract listener's attention and this will be done in ways that follow two conditions, in short these will have an overall influence on whether the said material may either manifest itself upon the foreground of our perception or recede away from us and be regarded as background. The main ways this happen can be summarized as follows:

(a) The sonic attributes of material that result from various activity of a certain parameters.
(b) The degree of movement of the material (which shall hereafter be explained so that it covers both conditions in greater detail).

The sonic attributes of material (a), based on my experiences as a composer, help to recognize a level of combination between the following parameters, traits that can contribute to determining which elements are perceived as being part of the sonic foreground.

Amplitude: We tend to hear louder sound as foreground and softer sound as background.

Frequency: We tend to here middle/high frequency sound as foreground and low frequency sound as background. For example, 500 Hz sound can be perceived as foreground than 50 Hz sound.

Pitch & Noise: We tend to hear pitched sound as foreground and noisy sound as background.

Duration: We tend to hear short sound as foreground and long sound as background.

Envelope: For example, we tend to hear sound with short attack time as foreground than sound with long attack time.

Reverb Settings: For example, we tend to hear dry sound as foreground than sound with a lot of reverb. Pre delay setting has also a big influence on foregrounding because it affects the attack characteristics of sound.

Recognizable pattern: We tend to hear sound with recognizable pattern as foreground and sound without pattern as background.

The above list has been obtained empirically, however, I am aware that I need to gain greater psychoacoustic proof of their presence, and this will be a subject for my future research.

The degree of movement (b), as it was mentioned in the previous section, tends to focus the attention of the listener on moving parameters rather than fixed parameters. The same goes for the properties of layers. When multiple layers are played simultaneously, our attention is directed to a layer that activates movement of some parameter. In other words, movement of parameters (e.g. pitch, rhythm, timbre, spatialization) pushes a layer to the forefront of a listener's attention. This fact can be considered in the following way: if a composer has an intention to place some layer in foreground, it is useful to apply techniques to give this layer a degree of timbral fluctuation. Especially, *short term, non-directional timbre movement* is useful for this purpose. In many cases it is also required to fix timbral movements of other materials in order to give stratified layers clarity and depth. Concurrent movements of timbre, happening in multiple materials, could also distract listeners' attention away from the layer that a composer wants them to focus on. This is the issue I encountered when I was composing *Colour Composition 2* and it will be discussed in the chapter 4-2. If a composer wants to give short-term timbre fluctuations to multiple layers simultaneously, the technique to introduce interaction between materials, which is mentioned in chapter 3-4, is useful, as it enables concurrent timbral fluctuations to be perceived as a holistic movement.

It is also interesting to use a compositional technique to push material that has an innate tendency to be regarded as part of the background into the foreground, and doing this by giving it fluctuation of timbre. This technique can be found in Monolake's techno track *Gecko*. From 4:00, the timbre of pitched sound is static. On the other hand, the quality of noisy sound is always alternating. The interesting point is that noise generally has a tendency to go to background, or attract less attention from listeners, compared with pitched sounds. If the timbre of noise in this piece would not change, or the timbre of both noise and the pitched sound changed, the noise would consequently be perceived as background. However, by fluctuating the timbre of only noise-based material, noise and pitched sounds obtain equal importance to audience.

Vertically stacked materials or layers sometimes introduce interesting phenomena. In some condition, superposed multiple materials make the boundary between layers blurry.

In variation IV from the second movement of Webern's *Symphonie*, Opus 21, repeating pitches are assigned to different instruments each time they are played. (Erickson, 1975) Therefore, the organization of layers would be different depending on whether they are to be understood from the perspective of pitch or timbre. This confusion between relationships of pitch and timbre, or ambiguous structure of associated layers, may result in giving listeners a unique aural experience.

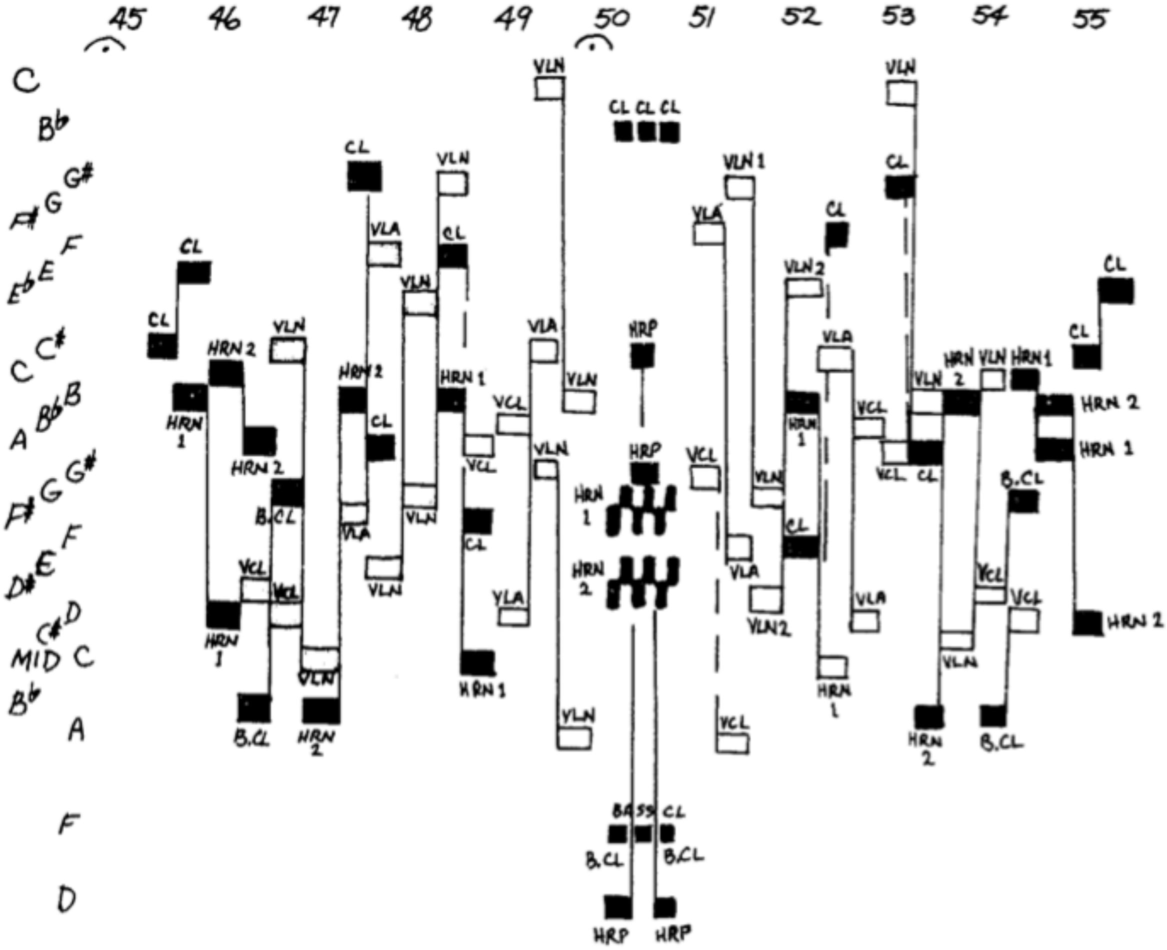


Figure 3-7-2: A graph of variation IV from the second movement of Webern's *Symphonie*, Opus 21, made by Ken Timm (Erickson, 1975, p. 116)

Wessel (1979) explains how the relationship between pitch and timbre affects the fusion and the separation of layers based on psychoacoustic research by Bregman and Campbell (1971), and Van Noorden (1975). Figure 3-7-2 shows the sequence of notes alternating between two different timbres. In this figure, "O" and "X" signify different timbres and in this example, when the timbral distance between "O" and "X" is small, we tend to listen to the repeating ascending pitch lines. However, when the timbral difference is enlarged, we tend to hear two interwoven descending lines, each with its own timbral identity. This phenomenon is called "melodic fission" or "auditory stream segregation" in psychoacoustic term.

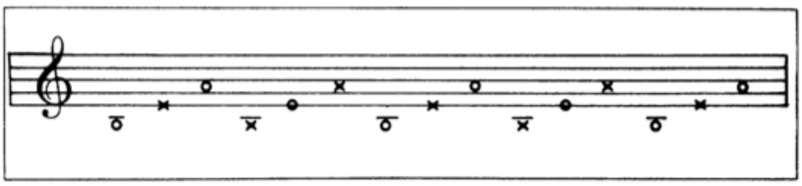


Figure 3-7-3: Example of melodic fission (Wessel, 1979, p. 6)

I find the confusion of layers, and the convergence/divergence of layers, caused by some pitch and timbral settings to be an interesting focus, one having great potential for compositional applications. The idea of gradually converging such layers was examined in the first part of my piece *Colour Composition 3* (to be explained in detail in chapter 4-3).

Relating to the merger of layers, Koenig suggested the concepts of "coalescing", "permeability", and "susceptibility" and although Koenig refers to the word "permeability" as being a borrowed term from Ligeti (personal communication, 23 April, 2015) his usage of it is presented in his own way.[2] To clarify, Koenig's idea of *permeability* refers to the predisposition of overlapping structures and their overall blending together. This is to suggest that if overlapping structures do not have contrast in terms of frequency, timbre, rhythm, and continuity (whether structure is made of long sound or groups of short sounds), a sense of coalescing can be said to have occurred. Therefore, according to Koenig, "If there is no such distinction, overlapping structures are in danger of coalescing and resulting in a uniform structure which can not be heard as having been put together from layers. We call such structures 'permeable' structures". (Koenig, 1965a, p. 14)

Susceptibility is a specific aspect of permeability. While permeability designates the ability of a sound structure to reveal another structure of a similar kind, whereas susceptibility makes them even more 'receptive' for the other, promoting the formation of a relationship and an increased sense of coherence. (personal communication, 23 April, 2015)

Koenig states, "I have always regarded this permeability as paramount for algorithmic composition" (Koenig, 1987, p.170). This phenomenon is used in his composition *Terminus* and *Funktionen*. In *Terminus*, the idea of coalescing is caused by superposing several transformations (Koenig, 1971) and the materials made from these transformations have common characteristics that can be understood to have descended from basic sound material. In other words, coalescing happens because superposed materials have a similar quality. According to Koenig, "If several transformations are superposed, their characteristics coalesce to give a new total impression whose complexity is above that of the individual layers". (ibid, p. 7)

In *Funktionen*, Koenig used sound materials generated by a single preset curve on the variable function generator. Control signals produced from the same curve were used to obtain derivations. Thus, all materials share some common characteristics coming from the variable function generator. Accordingly, they can be considered as permeable. In this piece, Koenig relied on the formative power of permeable structure, and since all the materials in structures are permeable, Koenig could leave the arrangement of sounds to chance in this piece. (Koenig, 1987)

Throughout the chapter, the subject of establishing relationships between materials by timbre movement is discussed. The synergy created by the superposition of related materials should now be explored more in my future compositions.

[2] Although Ligeti's idea of permeability deals with the merger of overlapping structures same as Koenig's idea, Ligeti concerns instrumental music and interval. According to Ligeti, permeability means that "structures of different textures can run concurrently, penetrate each other and even merge into one another completely, whereby the horizontal and vertical density-relationships are altered, it is true, but it is a matter of indifference which interval coincide in the thick of the fray." (Ligeti, 1964, p. 8) In Ligeti's discussion, permeable structure allows a free choice of intervals and impermeable structure does not. According to Ligeti, Palestrina's music is the example that has the lowest degree of permeability.

Chapter 4 Case Studies

In this chapter, three of my pieces will be explained in the chronological order of their creation. All of these pieces were composed during my research period in the Institute of Sonology, and each piece concerns different issues regarding the subject of timbre.

4-1 Flock

("1 Flock - stereo.aif" in the companion CD is the stereo mix of this piece)

Flock is the first piece of mine dealing with timbral transformations as a central subject. In addition, this is my first piece extensively using computer algorithms. It was composed before I developed BLOsc UGen in SuperCollider, thus I stuck to standard sound making methods. At the time, I was more interested in processing field recording materials than synthesis, this was because I had not developed sufficient techniques to achieve continuously changing timbre with standard synthesis methods and thought transforming concrete sounds was a better option to obtain sonic qualities with more microtime fluctuation. Barry Truax states, "At the most basic acoustic level, environmental sounds are much more complex in their spectral and temporal shape than most other musical material; synthesized sound in particular has been plagued by an artificial sound quality that has none of the corporeality of environmental sound" (Truax, 1996, p. 51). When I composed this piece, I had a similar viewpoint. Although I changed my mind later and came to believe that there are ways to achieve such complex spectra and temporal shapes with synthesis, however, I do still regard environmental sounds as being effective materials to bring sonic complexity into music.

Some of the initial ideas centered in this piece are as follows:

- Combine different timbre motions
- Develop music gradually, and achieve smoothly changing form and texture
- Make diverse sounds by processing field recording materials in various ways: filtering, phase vocoding, time stretching, chopping, and changing amplitude envelope
- Explore the various applications of tendency masks

The title "Flock" was chosen because various sounds of birds were used throughout the piece. The reason for choosing recordings of birds was that they contained many types of sounds. Some birds have pitched voices and other birds have noisy calls. A certain type of species makes sounds that resemble an impulse. Their fluttering sound can also be a good source for further processing. I was attracted by this sonic diversity because I wanted to obtain a variety of interesting materials by only manipulating a selection of field recordings.

It should be noted that when composing this piece, a graphical score was made in the beginning. (Figure 4-1-1)

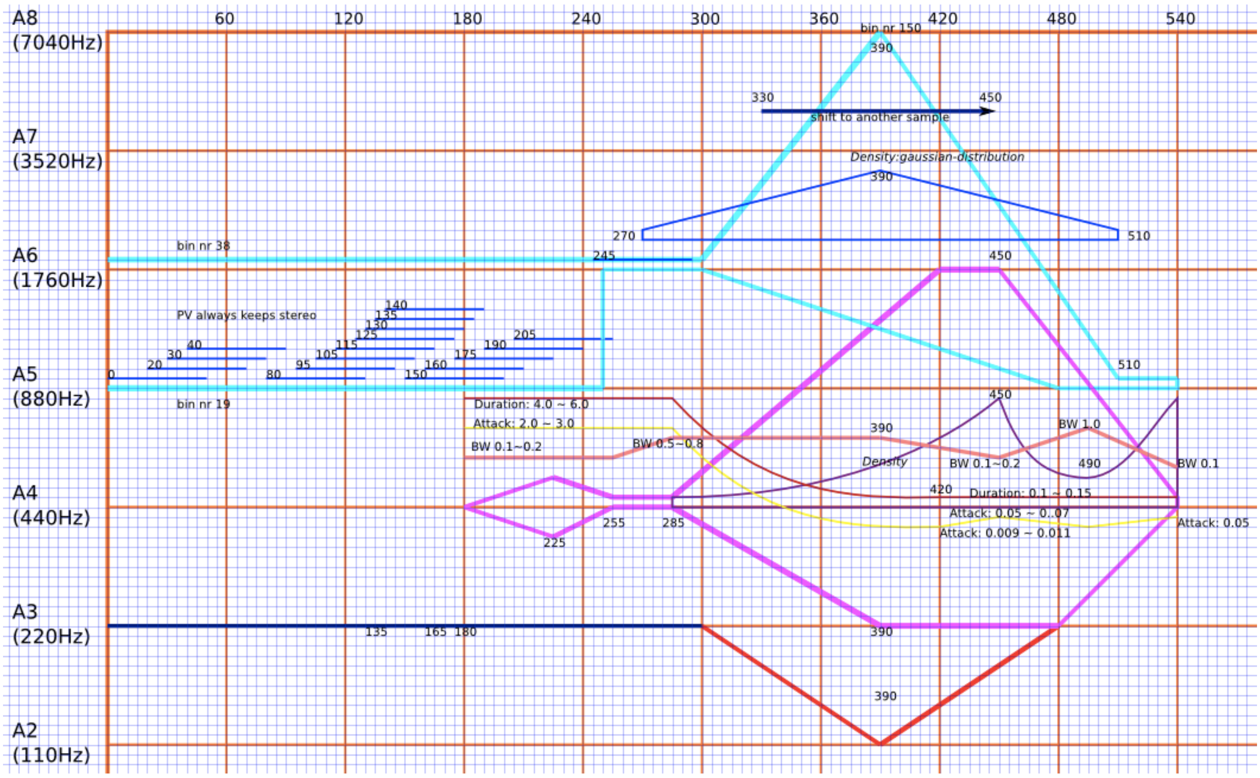


Figure 4-1-1: Graphical score of *Flock*

The horizontal axis of time is represented in seconds, while the vertical axis depicts frequency. Movements of pitch and other parameters are superimposed on this score. There are three frequency ranges specified each with tendency masks and each of them exhibiting a degree of expansion and contraction. As this score suggests, I wanted to make both tranquil moment (240 - 300 seconds from start) and climax moment (around 390 seconds from start).

Different sound processing methods were used for different registers. The higher register above 880 Hz was made with phase vocoding. The middle register between 220 Hz to 1760 Hz was composed of time-stretched sounds and granular synthesis. There was one octave of overlap (880 - 1760 Hz) between the high and the middle register. Lower register below 220 Hz was made of layers of transposed down samples. Those different processing methods gave each register different characteristics. However, they all use of heavily processed field recording materials and have continuously changing sound qualities that are in common. Detailed descriptions of each resister will now be explained.

1. Low register

The low frequency sounds made of layers of transposed down samples have a basic noisy quality, this contrasts with high and middle frequency sounds that exhibit clear pitches. Although noise contains energy not only in lower frequency but also in higher frequency, I categorized this sound as low register because it is the only part in the piece emphasizing this region of frequency. Sound files are played in 2 to 5 octaves lower than the original pitch. Accordingly, the duration is prolonged 4 to 32 times the length of the original recording. The transposition achieved by slowing down the playback speed is primitive, but nevertheless it remains an effective method to make a continuously moving sound. This method zooms in details and exaggerates the microtime sonic evolution.

In the beginning, only bird's voices are used in this register, but in the middle of the piece (5:04 -), the sound of fluttering joins. The flutter sounds are placed through a band pass filter, and its bandwidth is gradually narrowed. The filter's central frequency is around 600 Hz throughout the section. The reciprocal Q factor changes from 2 to 0.001. In the end, it is turned into very resonant sound with a clear pitch. This fluttering sound is especially important because the same material is used in the high register as the source of phase vocoding (partial isolation). The result is that the fluttering sound not only makes a connection between noise and pitch, but also establishes a connection between lower and higher registers.

2. Middle register

The intermittent long-sounds with pitch played between 2:15 - 3:12 are also made by stretching a bird's song, but this was done with a different signal processing method: the time-scale modification using phase vocoding. This was done by using pvanal and pvoc of Csound. FFT analysis file was made by pvanal program and it is read by slowly moving index using pvoc opcode. This method enabled time-stretching with or without frequency shifts. Therefore, the extreme time stretch, which is impossible to achieve by simple transposition became possible. The sound file was thus magnified a hundred times. However, the phase vocoding time-stretch introduced artifacts unique to FFT. One of the prominent artifacts was a metallic quality in the high frequency range, thus low pass filter in applied. Pitch shift was also applied and the sound was played approximately 3 octaves lower than the original recording.

From minute 3:00, another harmonic sound made of granular synthesis gradually becomes audible. The patch used to make these sound is shown in figure 4-1-2.

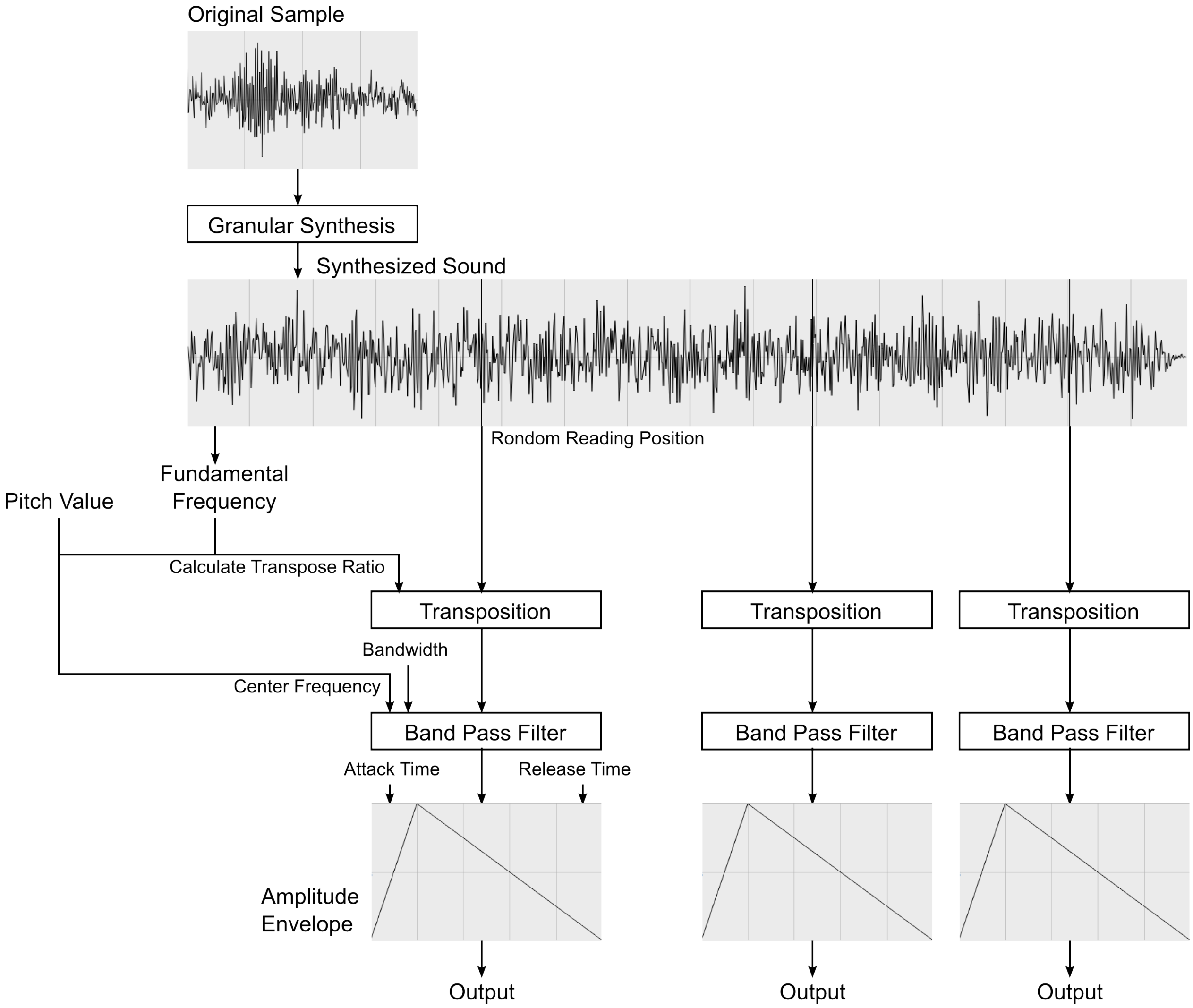


Figure 4-1-2: Patch to make middle range frequency sound

The granulated sample is bird's voice, again. The granular synthesis generates approximately 30 seconds of sound ("2 Flock - granular material.aif" in the accompanied CD). This sound has steady fundamental frequency with sufficient high harmonics. The advantage of granular synthesis in timbral composition is that it can produce sounds that have both a steady pitch and a lot of microtime fluctuation at the same time. This type of specific sonic quality is difficult to attain with normal oscillators. 30 seconds of generated sound are then loaded into a buffer. Whenever a sound is triggered, randomly moving index reads the buffer. Because of this randomized reading position, generated sound does not suffer from static quality. At the same time, precise pitch control is possible because the fundamental frequency of the sound, loaded onto the buffer is more or less constant. Additionally, the snippet of read sound is transposed based on the pitch sequence, to be achieved and the fundamental frequency of granular sound. (see Figure 4-1-2) After the transposition, a band pass filter is applied. The central frequency of band pass filter is then chosen, based on the same pitch sequence used to determine its transpositional ratio. Because of the rich higher partials that the sound loaded on the buffer contains, a band pass filter can create enough timbral variations by changing its bandwidth. In the end of the signal chain, an amplitude envelope is applied. As it was extensively discussed in chapter 1-3, parameters of amplitude envelope, especially attack, have great impact on resultant timbre. I have tried changing the shape of amplitude envelope of this patch over time.

The pitch sequence is made with tendency mask, with the pitch movement between 3:00 to 9:40 being shown on figure 4-1-3.

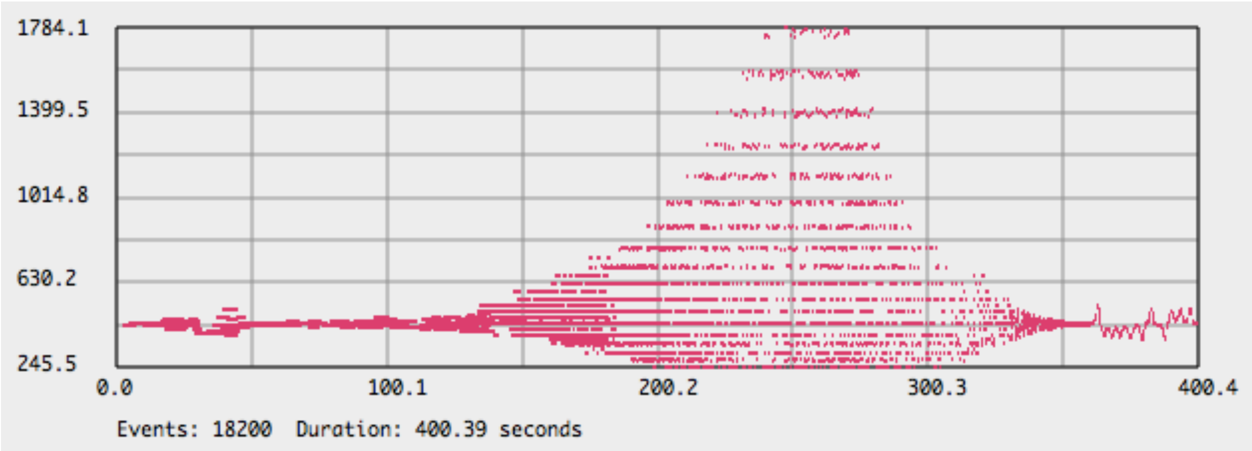


Figure 4-1-3: Plot of frequency values in middle register (3:00 - 9:40)

Once again, the horizontal axis is time and the vertical axis represents frequency. At first, frequency (midi note) values in between two narrow boundaries are chosen by a normal random function. Not only integer values but also floating point values are chosen as midi notes in this section. The duration of one note is quite long (12 to 18 seconds) during this section. However, the random sequence is gradually transformed into the whole tone scale concurrently with the widening of mask and the shortening of note duration (5:10 -). It can be achieved by rounding off the generated random midi note values to the multiple of 2. On figure 4-1-3, the middle section exhibiting clear parallel lines is when whole tone scale is played. The interval between two parallel lines corresponds to 2 semitones. From 8:10, the whole tone sequence goes back to the random sequence played between narrow boundaries. The intention of mine to make this behavior was to achieve not only the gradual change of timbre but also a gradual change of scale. In other words, the gradual shift between order and disorder in pitch dimension is intended.

In the concluding part of the piece (8:58 - 9:40), the pitch sequence exhibits a different gesture. Figure 4-1-4 is the plot of frequency values zoomed in to this section.

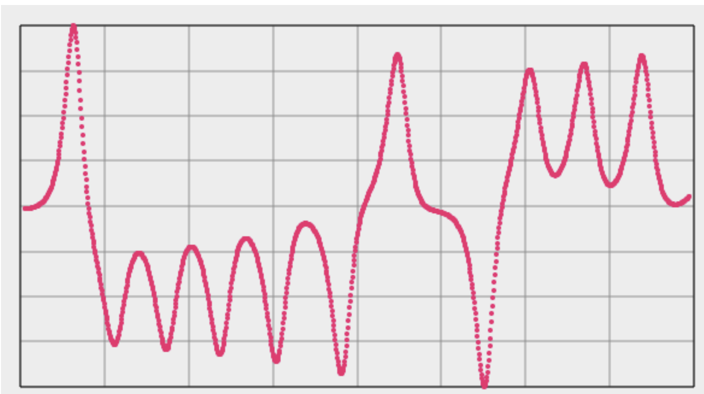


Figure 4-1-4: Plot of frequency values only in 8:58 - 9:40

This movement is made with Lorenz chaos equation in AC Toolbox. Agostino Di Scipio states, "In contemporary science, just like in ancient mythology, chaos stands for a highly dynamic situation, an ongoing and complex wave of turbulence between order and disorder, with several nuances in between. It's a dynamical process that may either bear order and form, or collapse" (Anderson, 2005, p. 11). As Di Scipio mentions, the useful application of chaos in music is to make a behavior that falls in between order and disorder. This is the intention that the chaos equation is used here. Both the middle part and the ending part are the exploration of boundary between order and disorder, but they take different forms.

Next, the timbral change in this register will be explained. In this part, the parameters that are most influential in timbre are the bandwidth of filter and the attack time. The changes of each parameter's value are shown on figure 4-1-5 and 4-1-6. Followed by the static state in the beginning (3:00 -4:15), the bandwidth starts widening. Once the bandwidth reaches a plateau state (4:45), the attack time starts decreasing. In the later part of the piece, the plot of bandwidth reveals two peaks with a valley in between. The valley moment corresponds to when the tendency mask for pitch is widest (7:20 - 7:30). In the concluding part of the piece, pitch is chosen by a chaotic equation (8:58 - 9:40), resulting in the bandwidth value being selected by the same equation.

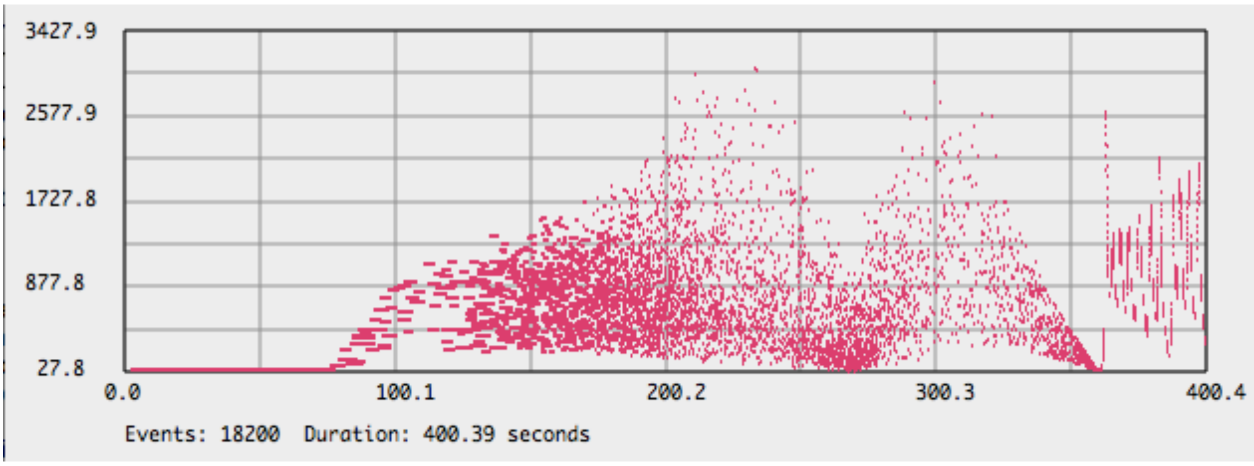


Figure 4-1-5: Plot of bandwidth in middle register (3:00 - 9:40). The bandwidth is specified in Hz difference between the upper and lower half-power points.

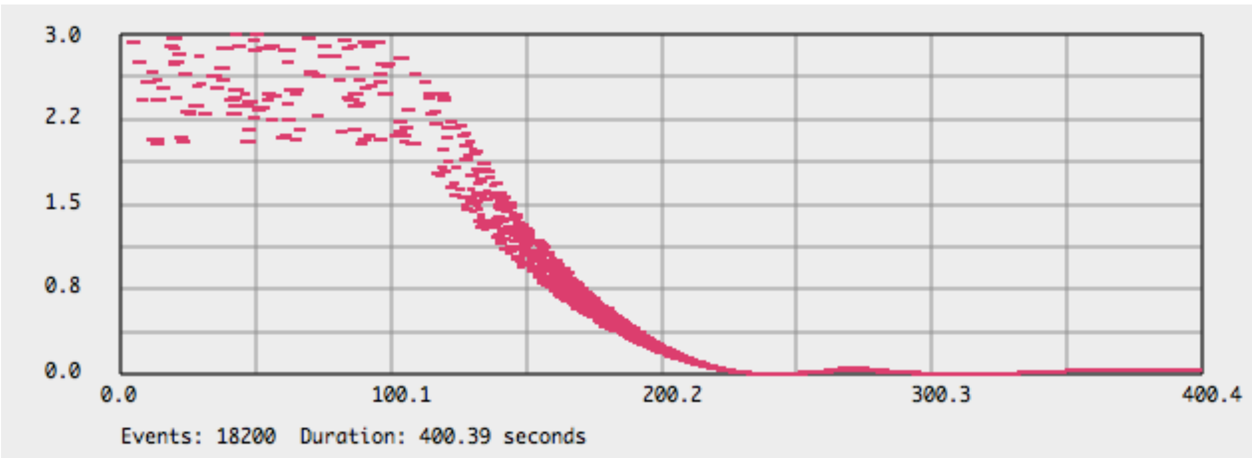


Figure 4-1-6: Plot of attack time in middle register (3:00 - 9:40). The attack time is specified in seconds.

3. High register

The exploration of timbral change using phase vocoder is intended in the high register. Csound opcode pvread, which reads one bin from an FFT analysis file, is applied. Each bin contains frequency and time-varying amplitude information. Multiple of bins can be considered as multiple of band pass filters (filter bank interpretation of phase vocoder) and complex filtering becomes possible by controlling the bins to be read. The sound quality of phase vocoder is merely an addition of amplitude-modulated sine waves when only a small number of bins are played. The quality of original sound file becomes evident when sufficient numbers of bins are played together. An amplitude envelope with long attack and release time (attack and release times are both around 15 seconds) is applied to each bin to avoid the abrupt emergence of a new sine wave whenever new bin is triggered. By slowly increasing the number of bins played together, the gradual transformation from sine wave quality to concrete sound quality occurs.

It is also possible to gradually morph one sample to another sample using phase vocoding. It can be done by making FFT analysis of multiple samples and changing the analysis file read by pvread opcode. This gives a totally different flavour of sample transition compared with amplitude cross-fading. In this piece, analysis files of four different bird's sounds are prepared and files to be read by pvread opcode are gradually altering. Snippets of four bird's sounds ("3 Flock - birds material.aif") are included in the accompanied CD in the order of appearance. In the beginning, calls from seagulls are used. From 5:15, a rhythmical swallow sound starts replacing the sustained seagull sound. From 6:00, the sound of swallow is slowly transformed into another bird's voice, which has a different intonation. This transition is intended to introduce rhythmical variations. From 7:43, a noisier fluttering sound takes over from pitched voice sample. As already mentioned, the same fluttering sound is used in the low register (5:04 -), thus a relationship between the high register and the low register is created. This part is a case of applying one processing method, namely phase vocoding, to various materials, which is mentioned in chapter 3-4-3, to achieve both the variations of sounds and the unity.

4. Reflections

During the composing of this piece, some issues concerning the use of field recording should be addressed.

a. Regarding the issue of identifiability of field recording sounds

In *Flock*, I did not want source materials that were processed to be too obvious. As it was already written, my interest in using field recording materials came from their complex spectral and temporal characteristics which enable me to obtain interesting sonic qualities by means of signal processing. In other words, I regarded field recording sounds as abstract source materials for further manipulation.

Regarding the field recording, there have been opposing standpoints in terms of the preservation of original context. On one hand, there is an attitude originated by Pierre Schaeffer and what Michel Chion refers to as 'reduced listening' (Chion, 1983). This attitude tries to strip context from the recorded sound. Schaeffer considered that dramatic sequences were unmusical because they referred listeners back to events and "the precondition to musique concrète was that samples be isolated not only from dramatic or anecdotal context, but also from their original musical context" (Palombini, 1993, p. 18). On the other hand, soundscape started by R. Murray Schafer emphasizes original contexts that recorded sound is situated, and there has been a confrontation between Schaefferian and Schaferian views with respect to musical creation (Lopez, 1997). My standpoint is completely Schaefferian side, because I feel the danger of making source material identifiable could limit listener's imagination, giving them the original context of the recording.

Another aspect considered regarding the identifiability of source sound in this piece was the peculiarity of the selected bird sounds. As previously mentioned, the reason behind the extensive use of bird calls was due to their sonic diversity. On the other hand, this piece made me realize that bird calls are actually tricky material to use, because they have been used in many musical pieces and can be perceived as cliché by some listeners. To avoid the use of bird's song being considered cliché, it was necessary to have heavily processed materials in order to give them as much fresh quality as possible. It also depends on the familiarity of sound if the usage of a particular material is regarded as cliché or not. For example, the voice of seagull is familiar to most listeners, thus I felt that intensive manipulations were required. On the other hand, the sound of swallow, which is used in the middle section of the piece, already had a unique characteristics to me in the original recording and therefore I did not feel that it was necessary to modify its identity.

When processing the familiar seagull's voice with phase vocoder (partial isolation), I tried to find the point where the seagull became recognizable, and discovered that only 5 or 6 bins are enough to make me notice that the source material is a seagull. This is because the amplitude fluctuation of bins plays a big role to make listeners realize the input sound. Before composing this piece, I had presumed that the spectrum had a singular dominant influence in timbral perception. In fact, the amplitude change is as important as spectral shape in timbral perception. The experiment of phase vocoding with a seagull's voice made me realize the necessity of handling amplitude carefully when making pieces focused on timbre.

In the section where the seagull's voice is used for the analysis file of phase vocoder, 7 bins were played together in maximum (around 2:30 - 3:00). I judged that if more bins are played at the same time and the quality of seagull's voice becomes explicit, the sound could be considered as a cliché by some listeners.

b. Suitable materials for partial isolation

Partial isolation by phase vocoder with noisy and rhythmic material is a good way of making unusual sounds. The spectrum of such materials is widely spread throughout entire frequency range, thus isolating bins makes big influence on resultant sonic quality. They have an ample potential to be sculpted in interesting ways. The fluttering sound used from 7:43 is an example of such a material. On the other hand, partial isolation of sounds that have a clear harmonic structure is more difficult to make an interesting result, because spectra of such sounds cannot be largely modified. This sonic characteristic also affects the point (the number of bins triggered) where the source materials become recognizable or unrecognizable. The one big reason that seagull's voice is easily identifiable, even if only a small number of partials are read, is that it has a clear harmonic structure. Thus, I felt the necessity to strictly limit the maximum number of bins playing together, in order to veil its identity. On the contrary, in the section the swallow's sound or the fluttering sound were used as inputs for the phase vocoder, I could play much more bins together without suffering from the problem of identifiability because those materials are noisy.

c. Another problem: "technology is heard"

Even if one could successfully hide the identity of field recording sound with signal processing, it might cause another problem: the processing technique becomes foreground. When I made a presentation about this piece at the Institute of Sonology, I explained my effort to hide the sonic identity of seagull's voice using phase vocoding. However, one participant commented that he listened to the phase vocoding technique instead of seagull's voice in this piece. This experience made me realize that listeners who know techniques tend to listen to techniques instead of focusing on sound or composition. Even in instrumental music, piano players tend to listen to other piano players' technique instead of the composition being played. This issue was already discussed in chapter 3-4-2 and here I argued that the best solution to this is not to hide the technology, but to make use of the fact that "technology is heard" in a composition. This attitude may be applied to the issue of identifiability of field recordings. In this piece, I tried to strip the identity of the seagull's voice by limiting the number of partials read by phase vocoder. However, I can now think of alternative and constructive approaches to positively use the fact that some field recording materials are easy to identify. One good example is *Le Présent composé* by Bernard Parmegiani. In this piece, a transformation of an actual door sound becomes a synthetic sustained tone by adding very long reverb. If the door sound was not identifiable, the surrealistic effect brought by this transformation would not be fully achieved. I also imagine that juxtaposing some familiar recorded material and synthetic sounds that have similar sonic qualities may create an interesting effect. Such techniques should be explored in my future composition.

d. Complexity made by interrelations between different layers

Although the strategy of applying different sound processing methods to layers occupying different frequency ranges was determined from the beginning, I found out during the composition that music becomes very boring if each layer is completely independent and there is no relationship between them. I felt the lack of complexity in the music when there were not sufficient inter-layer connections. This is the point when I obtained my interest in the subject of complexity and the relationships between materials. As a solution, I decided to use common field recording materials in different layers to give them relationships. For example, the fluttering sound was processed by a phase vocoder in the high frequency layer and the same material is transposed down in the low frequency layer. The sound of swallow is also heard both in the high and the low register. My interest in making relationships among materials was explored more extensively in my later pieces *Colour Composition 2 and 3*.

4-2 Colour Composition 2

("4 Colour Composition 2 - stereo.aif" in the companion CD is the stereo mix of this piece)

After the development of Band-Limited Oscillator UGen, I decided to make a series of compositions with it. I titled the series *"Colour Composition"*, which implies the exploration of timbre, or sound colour. *Colour Composition 1* is an etude, thus it has not been played in public and will not be discussed here. *Colour Composition 2* is the first concert piece of this series. When this piece was composed, my interest was in making various timbre movements with Band-Limited Oscillator and uniting such different movements into one single piece. To be more precise, the following issues are explored in this piece.

1. Fluctuating timbre
2. Seek the potential of Band-Limited Oscillator
3. Use different types of sounds: pitch/noise, synthetic/concrete, short/long
4. Connect different musical materials through timbral movement
5. Transform sounds with various ways
6. Examine the effect of density on timbre
7. Give richness and a lively quality to electronic sounds

The followings are the detailed explanations of the approaches taken to tackle those issues and the results obtained.

1. Fluctuating timbre

To make fluctuating timbre with Band-Limited Oscillator, it was necessary to take different approaches for short and long sounds.

a. Fluctuating timbre of short sound (1:08 - 3:20 and 7:41 - 8:36)

This sound was made by simply applying percussive envelopes to Band-Limited Oscillator. *loHarmonics* is fixed to 1. *numHarmonics*, *slope*, and *evenOddRatio* are randomized within the range of tendency masks every time a note is triggered. Tendency masks were also used to decide attack and release time. Figure 4-2-1 represents the plotted values of *frequency*, *slope*, and *evenOddRatio* in 1:08~3:20. It shows that the distribution ranges of parametric values became wider at the moment the density became higher.

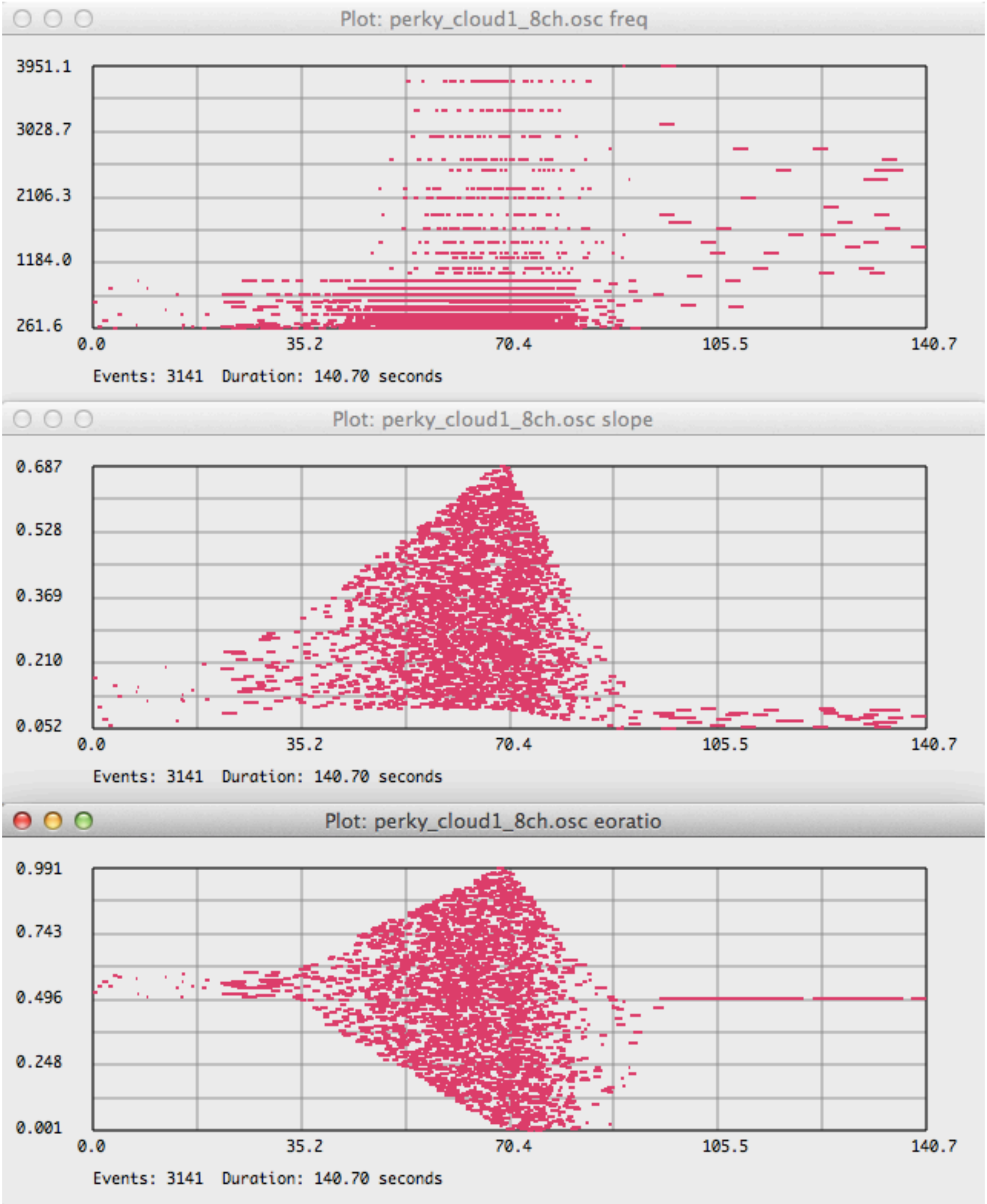


Figure 4-2-1: ***frequency*, *slope*, and *evenOddRatio* in 1:08~3:20 (from the top, plotted with AC Toolbox)**

b. Fluctuating timbre of long sound (1:08 - 5:40)

Fluctuating timbre of long sounds was obtained based on patches explained in chapter 2-4. *slope*, *evenOddRatio*, and *amp* values were modulated by low frequency noise (LFNoise). An amplitude envelope with sustain time, sine shape attack, and sine shape release was applied to each note. The range (the upper limit and the lower limit) and the frequency of low frequency noise, which was used as a modulator, were changed over time, thus the degree of timbral fluctuation was changed accordingly. Figures 4-2-2 shows that fluctuation of *slope* parameter got bigger (with wider modulator range) and more intense (with higher modulator frequency) at the end of section (horizontal axis: time, vertical axis: parameter values).

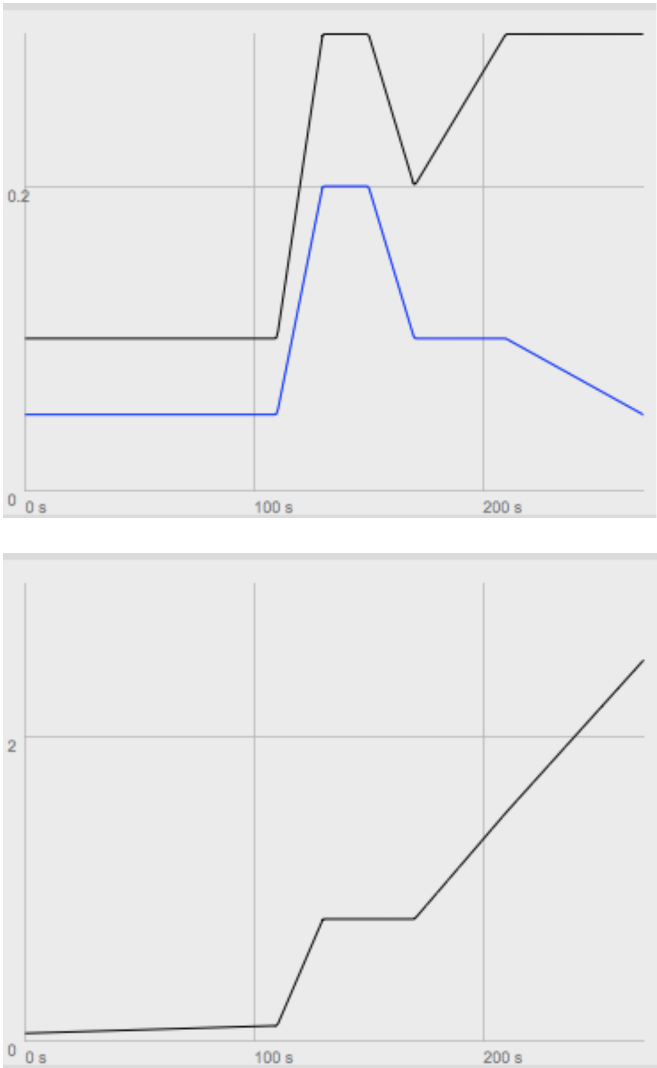


Figure 4-2-2: The change of upper limit and lower limit of low frequency noise applied to *slope* parameter (top)
The change of frequency of low frequency noise applied to *slope* parameter (bottom)

c. Fluctuating timbre of quasi-sustained sound with grains (4:08 - 6:50)

It is also possible to make fluctuating timbre with grains. By congregating small grains of sound made of Band-Limited Oscillator with different parameter settings, perceptual sound quality comes close to sustaining long tone whose timbre constantly moves.

The high pitch layer in 4:08-6:50 was made from quasi-sustained sound, made of grains of 5 to 7 seconds contained within a hanning envelope. Although this duration may be too long to be called “grain," I intentionally chose this term because of its principle’s resemblance to granular synthesis (granular synthesis typically uses grains of 10-100ms). Frequency is chosen with exponential distribution, of which threshold continuously decreases. *slope*, *evenOddRatio*, and panning position of each grain are determined by simple random distribution.

2. Seek the potential of Band-Limited Oscillator

Not only using Band-Limited Oscillator alone, the convolution of Band-Limited Oscillator and a field recording sound is also experimented. (4:57-8:34) More detail is explained later.

3. Use different types of sounds

Mixing different types of sound is an effective approach to obtain sonic richness in a composition. In this piece, the following combination of contrasting sound materials and contrasting sequences were used.

- Oscillator sound and field recording sound
- Harmonic sound, noisy sound, and impulse-like sound
- Short duration sound and long duration sound
- Short attack sound and long attack sound
- Tonal sequence and atonal sequence

4. Connect different musical materials through timbral movement

As explained in chapter 3-5, relating different sound materials through timbral movements contributed to both the consistency and the complexity of my own music. The following two methods were used to connect the materials contained within this piece.

a. Turning noise-based field recording sound into pitched synthetic sound using convolution
(4:57 - 8:34)

The convolution with Band-Limited Oscillator is used to interpolate contrasting sonic characteristics mentioned above (pitch/noise, synthetic/concrete). A noise-based field recording material, specifically a sound of insects, gradually obtained a synthetic quality with a clear pitch by decreasing *numHarmonics* of Band-Limited Oscillator.

b. Timbre side-chaining (This effect is most obvious in 1:08 - 1:35 and 2:42 - 3:15)

By introducing "timbre side-chaining", a technique explained in chapter 3-4-2, an interaction between short sounds and long sounds was created. The long sounds generated by Band-Limited Oscillator became brighter when short sounds appeared. To be more precise, *slope*, *evenOddRatio*, and amplitude of long sounds are modulated reacting to the amplitude of short sounds, which is the source of side-chaining. This is the case of "relation between one sound's amplitude and another sound's timbre" in chapter 3-4-2.

5. Transform sounds with various ways (2:42 - 6:05 and 4:57 - 8:34)

In chapter 3-4-3, two approaches for sound transformation were discussed. In this piece, the strategy for transforming field recording material is to use a single material and apply more than one processing methods. In 2:42-6:05, multiple copies of one field-recording placed through different speaker positions and with slightly different playback speeds are played together. The density gradually increased and the perceived timbre was changed accordingly. At minute 4:57-8:34, the same material was transformed by convolving with the Band-Limited Oscillator. Both sections were overlapped (from 4:57 to 6:05) to achieve a smooth transition from one to the other.

6. Examine the influence of density on timbre

Although this subject is not covered in the previous chapter, transforming timbre by changing density is an interesting subject. This approach has already been experimented with to some extent in the early electronic music produced in Cologne studio. For example, Koenig accelerated the speed of the impulse series above 16 Hz and made it audible as a rough sound in *Terminus*. (Koenig, 1971) However, in my piece, the influence of increasing density of different types of materials for the purpose of emphasizing and altering timbre is examined.

a. Percussive short sounds transformed into a "cloud" (1:08 - 3:20 and 7:41 - 8:36)

The density of short pitched sounds starting from 1:08 gradually increases and the similar gesture happens again in the ending part of the piece. As it has been already explained, each short sound is made of Band-Limited Oscillator and has slightly different parameter setting. When the density is reduced, notes are heard separately and timbral differences for each note are audible (around 1:40). This density increases when notes start overlapping and we hear the aggregation of small sounds as a "cloud". Timbre differences of discrete notes are not audible as separate entities. Instead, the cloud exhibits the texture that is full of microtime changes (around 2:15). I found that the timbral differences of successive short notes are audible when the delta time between notes is 0.1 second, but not audible if the delta time is 0.05 second.

The density change of short percussive sounds can also be considered as an envelope change. With high density, sounds are overlapped and the contour of amplitude envelope is smoothed out. Since amplitude envelope, especially attack, has significant influence on timbre, this change of envelope shape from percussive to smoothed contour affects listener's timbre recognition.

b. Rhythmic sound transformed into noisy sound (2:42 - 6:05)

Increasing levels of density of the field recordings used as source material is also a subject of experimentation in this piece. As it was already mentioned, a recording of insect sounds, exhibiting impulse-like rhythm, was used in this piece. This material was chosen because the transformation from discrete rhythm to a continuous noise-like quality was intended.

c. Harmony transformed into timbre (5:40 - 8:34)

The boundary between timbre and harmony is obscure. Erickson explains the ambiguities, relationships, and transformations concerning timbre, pitch, and chord using the following diagram. Other composers such as Tristan Murail even think that the concepts of harmony and timbre are theoretically same. (Murail, 2005) I can neither discuss their ideas in detail nor judge their validity, instead I simply would like to show the border between timbre and harmony, and how it is explored in this piece by changing the level of density. In turn, the ambiguous realm between harmony and timbre seems to be an interesting field to explore in composition, and developing the productive usage of this area is likely to be even more valuable to me than discussing whether harmony and timbre are different or not.

Figure 4-2-3: Erickson's diagram (Erickson, 1975)

The excerpt of this sound is included in the accompanied CD ("5 Colour Composition 2 - excerpt - glissand cloud.aif"). At first, discrete Band-Limited Oscillator sounds with a hanning envelope are heard. However, when a density increases and multiple instances of an oscillator sound start overlapping, they show how fusing effect containing discrete notes can be transformed into an inseparable sound, an enhanced "sonic complexity". The horizontal axis of figure 4-2-4 is time and the vertical axis is the onset frequency of each note. This graph shows that both the density and the duration of each note are increased over time. The densest moment is about 110 seconds after the start of the section and vast amount of notes are overlapped. However, it is presumed that the majority of listeners do not recognize so many notes that are played together because of the overall cohesion of the sound.

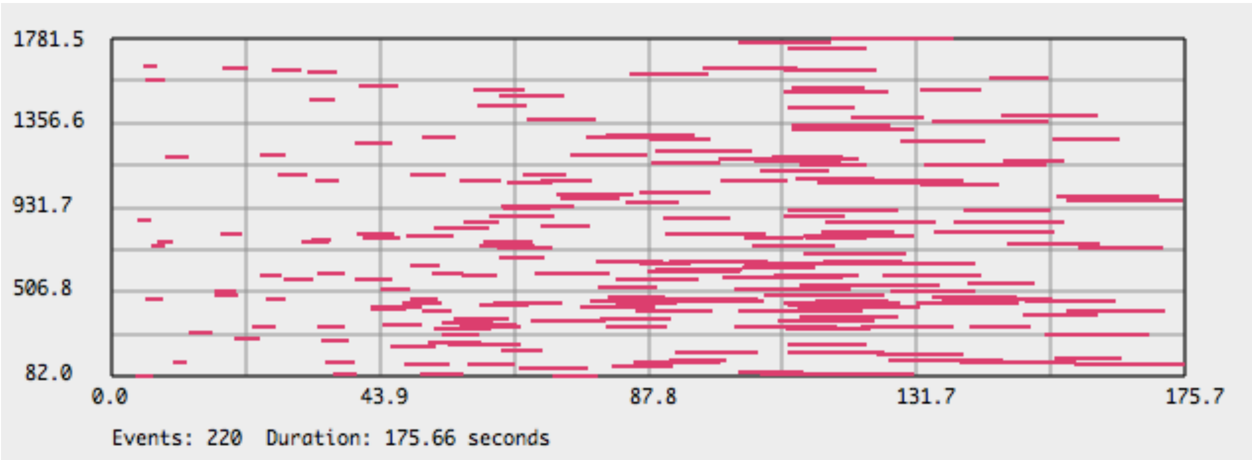


Figure 4-2-4: frequency of note onsets (plotted with AC Toolbox)

Pitch and timbre movements are also implemented in each note to give more microtime complexity to fused sound. From the onset to the release of each note, *frequency*, *slope*, and *evenOddRatio* are either increased or decreased. For example, the glissando will go up or down by a random amount, with a maximum of 20 % of the initial frequency value. The movement of *slope* and *evenOddRatio* are implemented in similar fashion (figure 4-2-5). I found out that even a single note can have interesting quality with the combination of glissando and timbral change. This experience convinced me of the validity of Koenig's idea that a single sound can provide the form for an entire composition of an electronic music. By adding up these moving notes, continuous and unpredictable spectral evolution is achieved.

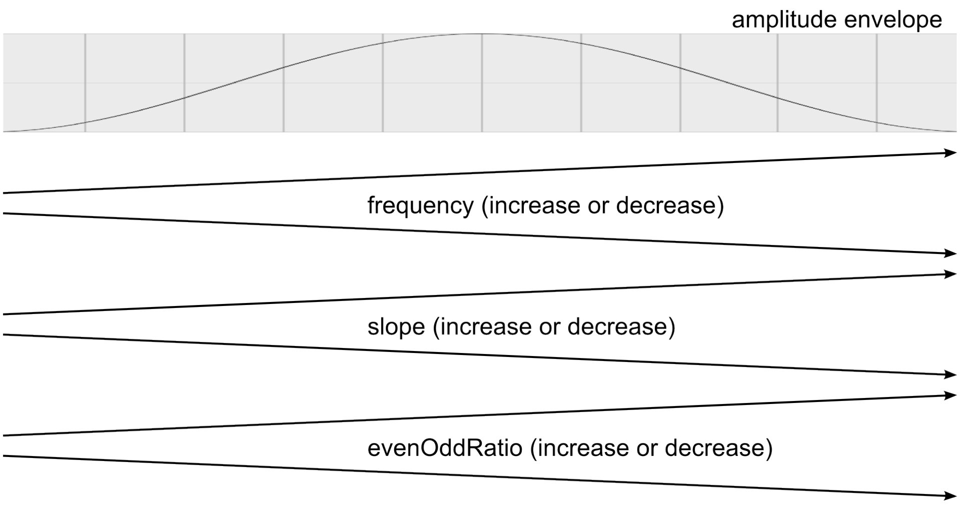


Figure 4-2-5: pitch and timbre movement of one note

7. Giving richness and a lively quality to electronic sounds (0:00-1:08)

The sound of major triad opening this piece is designed based on the question “why is the orchestra such a rich sound?”, this question was addressed in chapter 3-5-1. Of course my intention was not to simply imitate the sound of an orchestra, but to make a rich and fresh sound by combining the nature of a real ensemble with the advantage of electronics. In this part, 30 layers of Band-Limited Oscillator sounds are played together. Each layer is analogous to one string player of an orchestra. Each layer is made of a patch shown in figure 4-2-6.

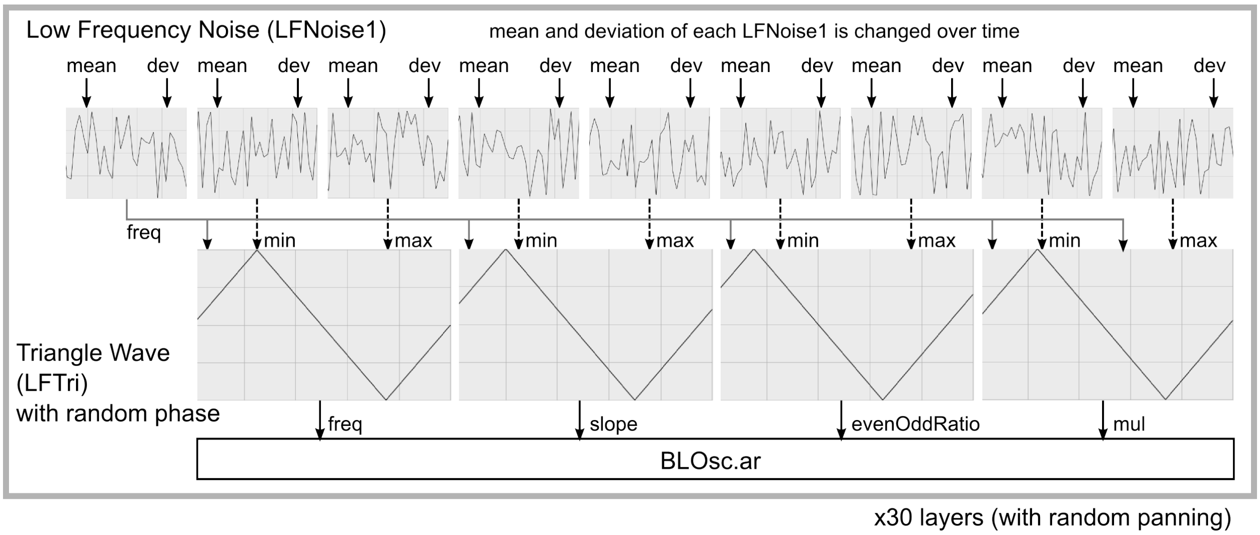


Figure 4-2-6: patch used in the opening section

Frequency, *slope*, *evenOddRatio*, and *mul* (amplitude) of Band-Limited Oscillator are modulated by different low frequency triangle waves. Frequency modulation makes vibrato, amplitude modulation makes tremolo, and *slope* and *evenOddRatio* modulation makes fluctuating timbre. Four triangle waves share the same frequency but have a different initial phase. Frequency, minimum value, and maximum value of the triangle waves are controlled by low frequency noise (LFNoise1). This means, the modulation of Band-Limited Oscillator is randomized and an unpredictable degree of microtime timbral change occurs. This is an implementation of short term timbre movement, discussed in chapter 3-2. The mean value and the deviation range of each LFNoise1 can be changed over time, using an envelope. The technique to determine parameter values by using random numbers that have mean values and distribution ranges was found in James Tenny’s Noise Study (Tenny, 1969), and this technique resembles the use of a tendency mask. Indeed, the same behavior can be achieved by determining the minimum and the maximum values of noise generators using tendency masks. This method enables combining short term, non-directional timbre movement with long term, directional timbre movement.

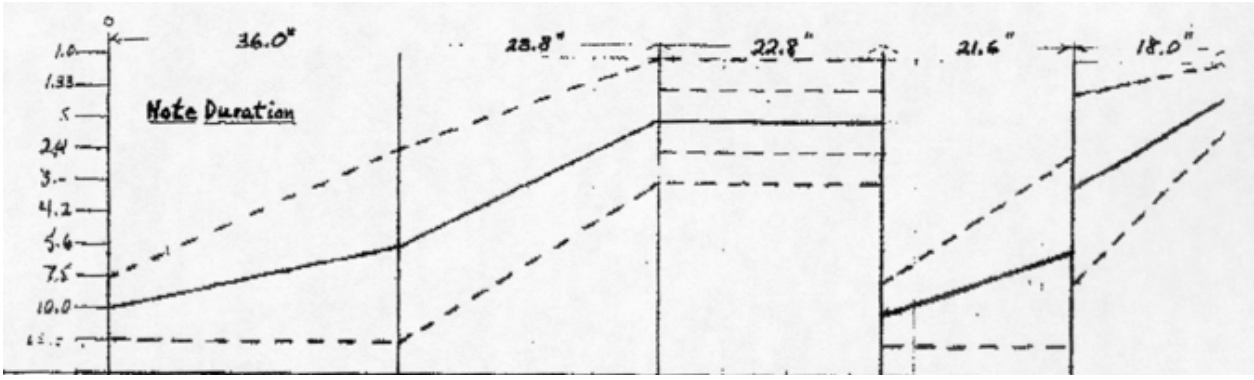


Figure 4-2-7: Example of parametric means and ranges found in James Tenny's Noise Study. Mean value is shown as a solid line and the range is between dashed lines. (Tenny, 1969, p.9)

The speeds of vibrato, tremolo, and timbre fluctuation are also controllable by regulating the leftmost low frequency noise on figure 4-2-6. The frequency of this noise generator gradually decreases and tremolo slows down accordingly. Precise rate control of vibrato and tremolo, and the mixture of short term and long term timbre movements is difficult to achieve with a real orchestra, but are easily implemented in computer program.

By layering this patch, rich moving sounds can be obtained. Randomized phases of triangle wave generators per layer resemble the fact that each player in orchestra has slightly different articulation. Randomized panning per layer is similar to the fact that each string player takes place at different position on a stage.

8. Reflections

a. Predictability in slow music

Although having and developing a slow musical evolution helps listeners to focus their attention on minute changes of timbre, it also brings the problem that the music can become predictable, due to a lack of surprising elements. The solution to this problem was explored in my next piece, *Colour Composition 3*.

b. Short sounds vs. long sounds

Sequences of short sounds sometimes distract listener's attention away from the inner changes of simultaneously played long sounds. In other words, listeners tend to hear short sounds as foreground and long sounds as background. This issue has already been addressed in chapter 3-7, and to exemplify this phenomenon, I have included two tracks in the companion CD. Track 6 is the excerpt from the final version of the piece (3:20 - 5:44). Track 7 is the version I eventually decided not to use. In this alternative version, short pitched sounds are present that the final version does not contain. The amplitude envelopes of short sounds gradually transform from percussive shape to hanning window like shape with slower attack time. This transformation is intended to bridge the "perky-cloud" section before 3:20 and "glissando" section after 5:44. However, I found out that the existence of short sounds distracts the listener's attention away from the long sounds where timbre is actively fluctuating. Since there is no other section in this piece that listeners can fully enjoy the timbral fluctuating of long notes, I have decided to remove short sounds from this part.

In short, I feel it is advisable to make a section dedicated to long sounds without distracting short sounds if a composer wants listeners to enjoy the timbral evolution of long notes. This reflection also made me realize the danger of adding up too many layers especially if materials are full of microtime sonic evolutions. It appears that less layers give listeners clearer view of minute timbral changes in each material.

4-3 Colour Composition 3

("8 Colour Composition 3 - stereo.aif" in the companion CD is the stereo mix of this piece)

In this piece, my intention was to make an interplay between two timbral poles: pitch and noise, with bell-like sounds used throughout this piece acting to bridge these two poles. Although there are clear resonant peaks in the spectrum of the bell sounds, those peaks are not evenly placed and the harmonic structure is unclear. Because of this fact, the bell sound perceptually seems to fall in between pitch and noise. As it was already explained in chapter 3-4-2, the use of bell as a way to bridge pitch and noise can be seen in the third movement of Webern's *The Five Pieces for Orchestra*, Op 10.

To achieve a meaningful interplay between pitch and noise, various "timbral counterpoints", explained in the earlier chapter, were implemented in this piece. Both vertical (occurring at the same time) and horizontal (occurring at the different moments of time) networking of different musical materials through pitch, rhythm, and timbral relationships were explored.

Exploration of the usage of rhythmic materials, which was not dealt in my previous pieces, was another important subject of this piece.

This piece can be divided into six sections and each part exhibits different kinds of interaction between pitch and noise.

1. The first section (0:00 - 2:14)

The following elements are used in this section:
SOUND 1-A: Synthetic bell sound (with prolonged tail)
SOUND 1-B: Percussive band-pass filtered noise
SOUND 1-C: Band-pass filtered pulse

SOUND 1-A (synthetic bell sound) begins the piece. This sound was made using SuperCollider's Klank UGen (a bank of fixed frequency resonators). The harmonic structure of this sound is related to pitches for next two notes (SOUND 1-B: filtered noise with very narrow band). The center frequencies of these two filtered noises are equal to midi note number 60 and 96. The first bell sound contains two spectral peaks corresponding to note number 60 and 96. Of course note 60 and 96 are not the only resonant frequencies of this bell sound. In fact, there are total 25 spectral peaks. Those resonances are heard throughout the rest of section as multiple of sustained sine tones. Listeners hear these sine tones as a prolonged tail of the bell. The amplitude of this tail of the bell is modulated by the amplitude of filtered noises (SOUND 1-B), using side-chaining. In this way, a relationship between the bell sound and filtered noises occurs.

SOUND 1-B constitutes two layers, starting from different pitches (60 and 96), different spatial positions (left and right), and different meters (4/4 and 3/4). However, these two layers gradually merge into one. Pitch ranges (tendency masks) of two layers become wider and gradually overlapping. In addition, the bandwidth of filters is also widening and the sense of pitch is getting lost. The change of rhythm (the density increase) incorporates the sequence of quarter notes and the sequence of triplets into a unified sequence by the end of the section.

The pitch selection principle inside tendency mask is based on brownian motion. Different from normal brownian motion, which can take any step value between specified lower and upper limits, step sizes here only take either ±3 or ±4. This number corresponds to the meter (3/4 and 4/4). More importantly, ±3 or ±4 steps brownian motion makes pitch sequences whose qualities fall in between tonal and atonal. If successive two step sizes are +3 and +4, it corresponds to the arpeggio of minor triad. +4 and +3 steps corresponds to major triad. If these step sizes are taken, a tonal quality is emerged from the sequence. On the other hand, the succession of +3 corresponds to diminished triad and the succession of +4 corresponds to augmented triad. If successive step sizes are +4 and -3, for example, the interval between the first note and the third note is minor second. These dissonant interval patterns do not contribute to the emergence of tonality. This 3/4 steps brownian motion is employed throughout the piece.

SOUND 1-B (filtered noise) and SOUND 1-C (filtered pulse) show opposite timbral movements. The bandwidth of SOUND 1-B widens and the transformation from a pitch specific quality to a noisy quality occurs. The bandwidth of SOUND 1-C narrows and the transformation from a noisy quality to a pitch specific quality occurs. This is an attempt of making timbral counterpoint. On the other hand, SOUND 1-B and SOUND 1-C share the common movement in terms of density, which increases over time. This shared movement makes the connection of SOUND 1-B and SOUND 1-C stronger.

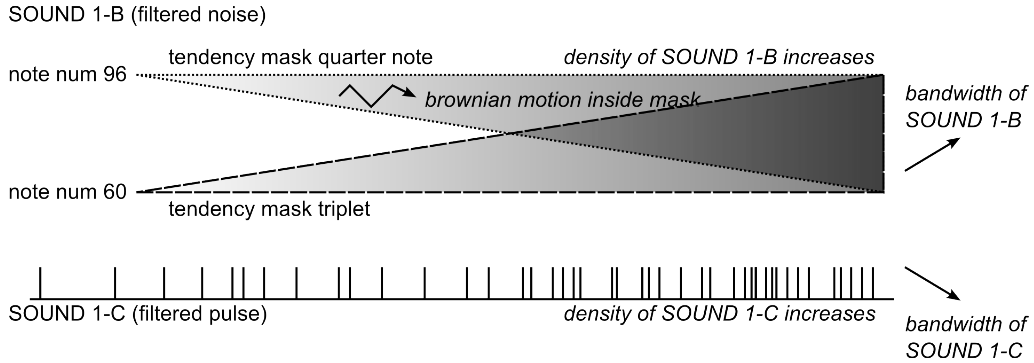


Figure 4-3-1: Model (relations between sounds) used in the first section

The spatialization of filtered noise and filtered pulse is determined to reinforce the merging of layers and to clarify the timbral counterpoint. Figure 4-3-2 and 4-3-3 show how the position of filtered noise and filtered pulses occur at the beginning and at the ending of the section. The position of one hit of percussive noise and pulse is determined randomly within the arrow. Therefore, this arrow can be considered as the tendency mask applied to panning position. In the beginning, filtered noise with note number 60 comes out from front-left speaker and note 96 comes out from front-right speaker. This positioning emphasizes the existence of two different layers. The tendency mask of each layer gradually becomes wider, overlapping, and eventually covering the front 5 speakers. Two layers do not exhibit any spatial difference by the end of section and it helps the merger of layers. On the other hand, filtered pulses are heard from rear 3 speakers. As a result, the timbre of front speakers transforms from a pitch specific to noise-based quality and the timbre of rear speakers shows the opposite transformation. The separated playback position of noise and pulse is therefore intended to make opposing timbral movements comprehensible to an audience.

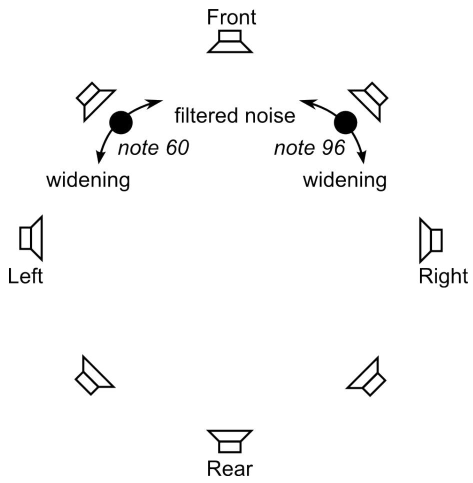


Figure 4-3-2: Spatialization at the beginning of the first section

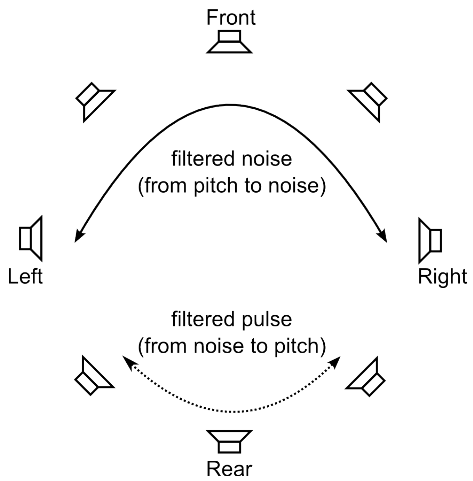


Figure 4-3-3: Spatialization at the ending of the first section

2. The second section (2:14 - 3:44)

This section is composed of the following four types of sounds:
SOUND 2-A: Percussive band-pass filtered noise continued from the previous section
SOUND 2-B: Percussive Band-Limited Oscillator plus percussive band-pass filtered noise
SOUND 2-C: Band-pass filtered pulse
SOUND 2-D: Recorded church bell sound processed with phase vocoding (partial isolation)

SOUND 2-A and SOUND 2-B are played alternately. In the beginning of the section, only SOUND 2-A is heard, but the proportion of SOUND 2-B gradually increases, and SOUND 2-B completely takes over by the end of the section. SOUND 2-A and SOUND 2-B share the same pitch sequence based on 3/4 steps brownian motion, resulting in a melodic continuity.

To make SOUND 2-B, Band-Limited Oscillator and filtered noise are played together. The oscillator sound and filtered noise share the same percussive envelope. The frequency of oscillator and the center frequency of filtered noise are also same. My intention to combine oscillator sound and filtered noise is to make a transformation between pitch and noise possible, by changing the amplitude balance of two sounds. Transformation between pitch and noise in the previous section was done by changing the bandwidth of filter. Thus, the transformation in the previous section and the transformation in this section have different flavors.

In this section, the merger of two layers is explored experimented again. Respective two layers are made of SOUND 2-A and SOUND 2-B. The merger of layers occurs in a different way from the previous section. The bandwidth of SOUND 2-A gradually narrows (rq = reciprocal Q factor changes from 1.0 to 0.05 throughout the section) and the sound gains a quality of pitch. The level of oscillator for SOUND 2-B gradually increases and that of filtered noise gradually decreases, thus it becomes more pitch specific quality, thus the timbral movement similar to that of SOUND 2-A occurs. SOUND 2-A and SOUND 2-B are clearly different in the beginning of the section, but sound qualities come closer by the end of the section, as a result "timbral merger" occurs.

Same as the previous section, noise and pulse show opposite timbral movements. Contrary to SOUND 2-A and SOUND 2-B, the bandwidth of SOUND 2-C (filtered pulse) gradually widens and the degree of noise increases. However, different from the previous section, the triggering timing (beat) of percussive noises and pulses are now synchronized. Noise and pulse have stronger connection than the previous section because of this rhythmical synchronicity.

This section also uses bell sounds (SOUND 2-D). However, a field recording bell sound processed with phase vocoder is used instead of the synthetic one. Similar to synthetic bell sounds in other sections, this sound is used to fill the gap between pitch and noise. Meanwhile, the intention to give "atmospheric character" to this section made me use the concrete material. "Atmospheric character" I mean is the result of natural reverberation and environmental noises contained in the original recording.

When SOUND 2-A is triggered, new bin for SOUND 2-D is triggered. When SOUND 2-B is triggered, no bin is triggered for SOUND 2-D. Although the synchronization between SOUND 2-A and SOUND 2-D is not perceptually obvious, this system still makes some sonic connection between the synthetic sounds and the field recording sound. When SOUND 2-A is denser than SOUND 2-B, more and more bins are triggered, resulting in the clearer field recording quality. When SOUND 2-B is denser than SOUND 2-A, the number of bins gradually decreases. Accordingly, the field recording quality becomes less obvious and the resultant sound approaches to an amplitude-modulated oscillator-like quality.

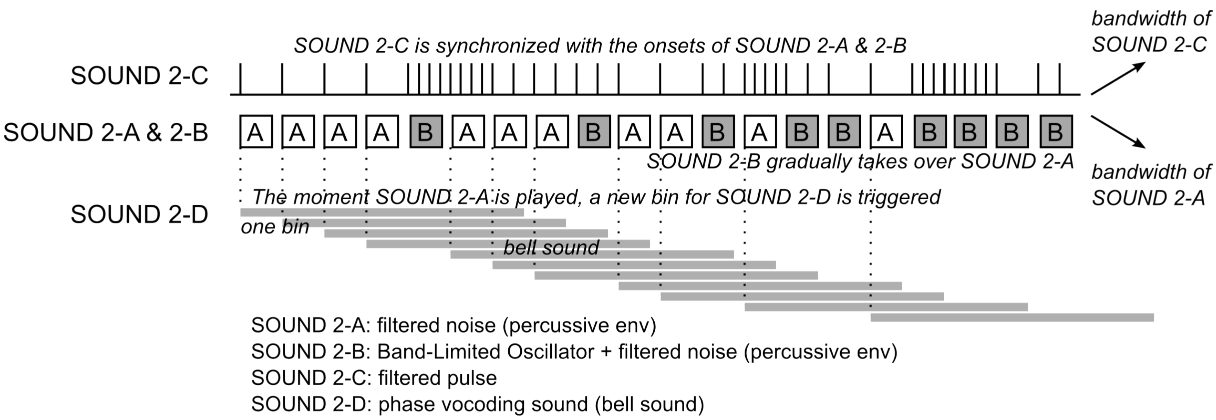


Figure 4-3-4: Model (relations between sounds) used in the second section

3. The third section (3:44 - 6:56)

This section is aimed at exploring interactions between noise and pitch in long sounds. This section is composed from the following sounds:

SOUND 3-A: Long lasting band-limited noise whose cutoff frequencies are modulated. This sound occupies high and low frequency ranges.

SOUND 3-B: Band-Limited Oscillator plus filtered noise with envelope of sinusoidal shape attack and release. This sound fills the middle frequency range.

SOUND 3-A (band-limited noise) is made by convolving white noise and BLOsc2. Both the low and the high cutoff frequencies are moving all the time, thus it becomes "fluctuating band-limited noise". This is achieved by modulating *loHarmonics* and *hiHarmonics* parameters of BLOsc2 with low frequency noise (LFNoise1). These parameters determine the low and high cutoff frequencies (bins) of band-limited noise.

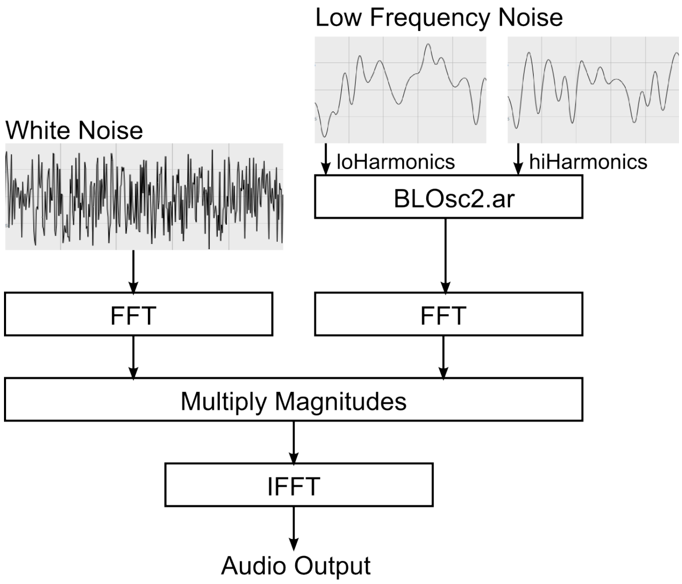


Figure 4-3-5: Signal flow to make fluctuating band-limited noise

The center values of two LFNoise1 are determined by tendency masks. In other words, tendency masks determine the mean values of low and high cutoff frequencies of a fluctuating band. Two band-limited noises are played simultaneously and each of them fills the high register and the low register. Both noises last more than 4 minutes. Tendency masks for two noises are basically widening. However, there is an empty frequency range between these two noises. My plan was to combine the high frequency noise and the low frequency noise, sandwiching SOUND 3-B occupying the middle frequency range, this was because it was intended that they use all the frequency ranges equally within this section.

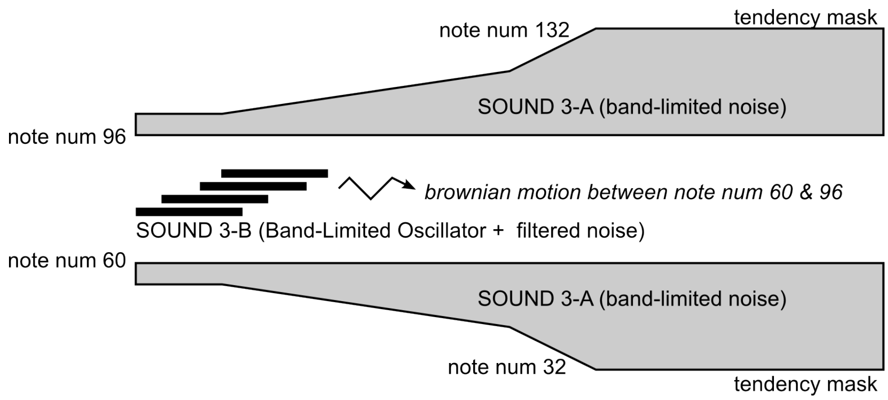


Figure 4-3-6: Musical development in the third section

SOUND 3-B resembles SOUND 2-B, but the amplitude envelope is different. While SOUND 2-B uses percussive envelope, SOUND 3-B uses sinusoidal attack and release shapes. There is a sustained time in between attack and release. This means one note can be very long, and the transdformation from pitched sound to noise-based sound within a single note is possible. The frequency of SOUND 3-B is again chosen by 3/4 steps brownian motion. The one instance of SOUND 3-B lasts about 60 seconds (including attack and release time. Within this duration, various parameters are modulated. The modulated parameters are shown in Figure 4-3-7. In general, one sustained sound increasingly gains the quality of pitch (the level of oscillator sound increases, the level of filtered noise decreases, and the bandwidth of filter narrows), brighter (*hiHarmonics* parameter increases), and actively fluctuating (the deviation ranges of *slope* and *evenOddRatio* parameters increase towards the release of the note. My intention of implementing this gesture in each note is to make the pitch specific quality of middle register sounds gradually appear in between the high and the low register fluctuating noises.

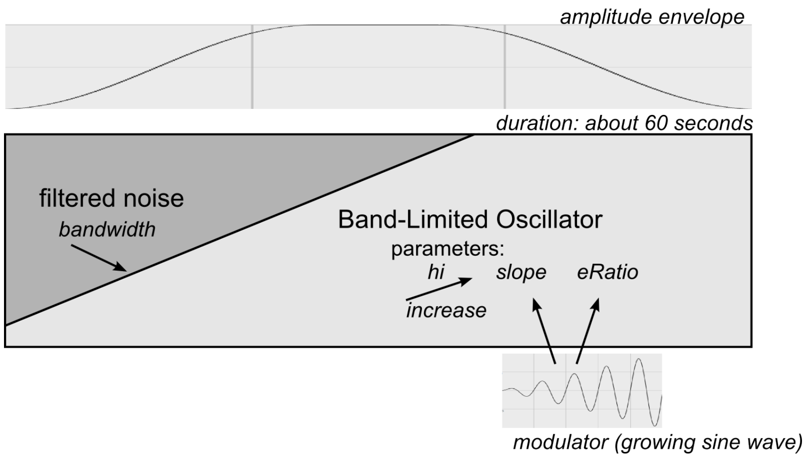


Figure 4-3-7: Evolution of one instance of SOUND 3-B

4. The fourth section (6:56 - 8:39)

This part plays a transitional role from the third section to the fifth section, and it is composed of the following sounds:

SOUND 4-A: Percussive band-pass filtered noise

SOUND 4-B: Percussive Band-Limited Oscillator plus percussive filtered noise

SOUND 4-C: Bassy Band-Limited Oscillator sound of which parameters are modulated responding to side-chain input
SOUND 4-D: Synthetic bell sound

SOUND 4-A (filtered noise) uses a SynthDef, which already appeared in section 1 and 2. The reason for repeatedly using the same SynthDef is to build connections among multiple sections, though the release time in this section is longer than previous sections. This sound also has the perceptual similarity with the next section's cabasa sound. This noise sound becomes increasingly present by widening the bandwidth of the filter and increasing the amplitude.

SOUND 4-B is basically same as SOUND 2-B, with the difference that from SOUND 3-B is only the shape of amplitude envelope. In this section, frequency is fixed to note number 96. This corresponds to the higher pitched sound played at the beginning of the piece. On the other hand, the fundamental frequency of SOUND 4-D (synthetic bell sound) is equal to note number 60, which corresponds to the lower pitched sound heard at the opening. It is intended to strengthen the unity of the piece by using same pitches (60 and 96) repeatedly in different sections. SOUND 4-B keeps triplet rhythm. The combination of note 96 and triplet is also the reprise of the first section. Timbral motion emphasizes this rhythm (Figure 4-3-8). There is a "balance" parameter in this synth and it determines the amplitude balance between Band-Limited Oscillator and filtered noise. Value 1.0 means only the oscillator sound is heard and 0 means only the filtered noise is heard. The balance parameter cycles the following value: 0.9, 0.2, and 0.05. Thus, timbre keeps changing with the following pattern: "pitch", "in between pitch and noise", and "noise". In synchronization with the timbral movement, the speaker position cycles the following: front-left, front-center, and front-right.

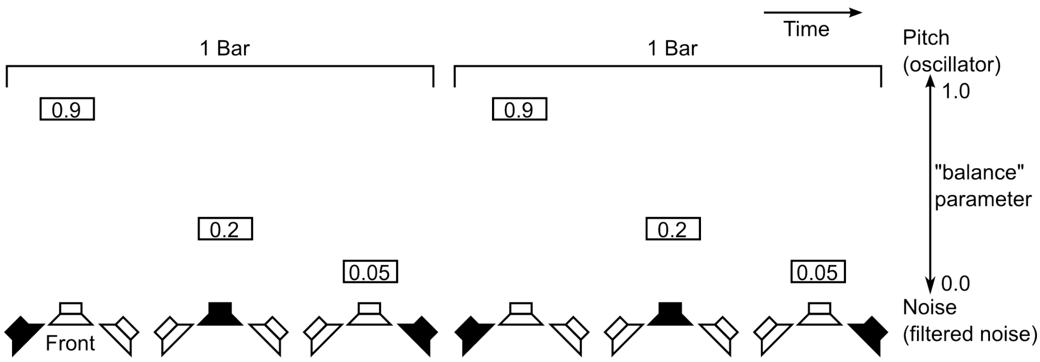


Figure 4-3-8: Timbral movements and spatialization of SOUND 4-B

This is an example of short term timbral movement, as it was explained in chapter 3-2 that is now being combined with panning movement. The purpose of the timbral fluctuation here is not only to give a lively quality to the sound but also to give rhythmical punctuation. In addition, the amplitude of SOUND 4-B is used as the source of "timbre side-chaining", which modulates SOUND 4-C.

SOUND 4-C is made with Band-Limited Oscillator and fills the lower frequency range of this section. Timbre side-chaining affects *slope* parameter. When SOUND 4-B is heard, the slope value is increased and the timbre becomes brighter. By this way, the timbral connection between SOUND 4-B (high frequency sound) and SOUND 4-C (low frequency sound) is created. The source of side-chaining is "pre-fader" amplitude of SOUND 4-B. At the end of the section, SOUND 4-B is faded out but the effect of side-chaining remains because of this "pre-fader" setting.

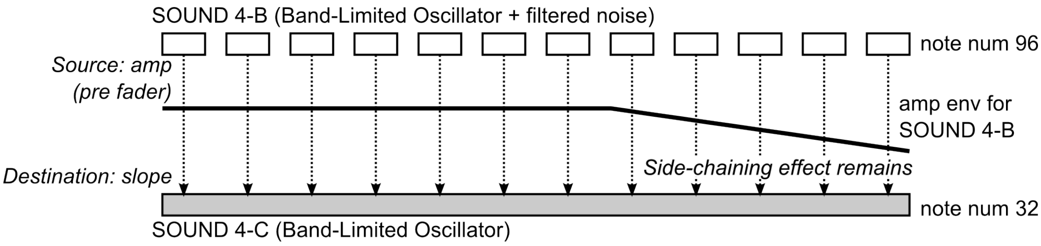


Figure 4-3-9: Timbre side-chaining used in the fourth section

SOUND 4-D is made with Klank UGen. The fundamental frequency of this sound is fixed to note number 60, but the characteristics are gradually changing by deviating frequencies of higher partials, amplitudes, and decay times. At first, frequencies of higher partials are more or less evenly spread, thus Klank UGen generates oscillator-like sounds rather than bell-like sounds. Towards the end of this section, the overtone structure becomes increasingly randomized. Again, the bell sound is used to bridge the pitched material (SOUND 4-B) and the noise-based material (SOUND 4-A) (vertical connection). Additionally, this sound performs the role of bridging this section and the next section (horizontal connection).

5. The fifth section (8:39 - 12:42)

This section is composed of the two types of sounds:
SOUND 5-A: The pad sound made with Klank UGen
SOUND 5-B: Percussion samples

Percussion instruments are appropriate materials for the main subject of this composition: as they articulate an interplay between noise and pitch. Different percussion instruments have different degrees of noisiness. Some percussion instruments exhibiting a clear pitch, however, by using various percussion instruments, a wide range of sound from noise to pitch can be expressed.

SOUND 4-D (bell sounds) is transformed into SOUND 5-A (pad sounds). In the previous section, the excitation input to the resonant filter bank of Klank UGen is impulse, thus bell like sound is generated. In this section, the excitation input is brown noise, thus sustained sound is generated. In the beginning of this section, the frequencies of Klank resonators are randomly dispersed. However, they are gradually converged to the multiples of fundamental frequency. Therefore, the sound comes closer to simple oscillator-like qualities heard at the end of section, in contrast to the timbral movement of bell-like sounds in the previous section.

Percussion (SOUND 5-B) used in this section are snare drum, cabasa, hi-hat cymbal, ride cymbal, triangle, and xylophone. Snare drum and cabasa make noisy sound and snare stick has decayed impulse-like quality. Triangle and xylophone make clear pitch. Hi-hat cymbal and ride cymbal make sounds in between noise and pitch.

Band pass filter is applied to each percussion sample. By randomizing the center frequency and the bandwidth of filter at each moment a sample is triggered, the timbre of percussion changes. Along with the slightly randomized amplitude and pitch (playback speed of sample in this case), this short term timbre fluctuation gives lively and rather humanized quality to this percussion sequence.

The key instrument of this section is xylophone, which keeps triplet rhythm. Triplet rhythm is taken over from SOUND 4-B, though tempo in this section is faster than that of previous section. Additionally, timbral movement and spatial movement reinforce the connection with previous section. Xylophone samples are played together with percussive filtered noise. By controlling the amplitude balance of xylophone and filtered noise, listeners can perceive how the noisiness of the xylophone changes. With higher balance value, filtered noise is less present. The balance parameter cycles the following value: 0.05, 0.3, and 0.9. The speaker position cycles the following: front-right, front-center, and front-left. This movement is opposite to the movement of SOUND 4-B (Figure 4-3-10). By implementing these timbral and spatial counter movements, a horizontal relationship is created. A long term timbre transformation is also implemented here. The bandwidth of filtered noise continuously decreases, thus the noisiness, especially of the first beat, is gradually attenuated over time.

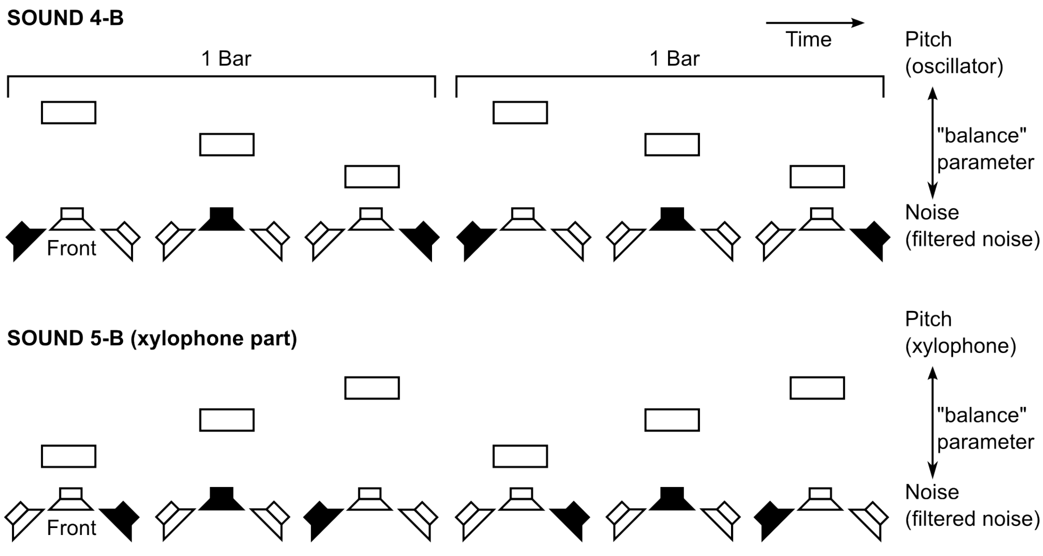


Figure 4-3-10: Timbral and spatial counter movements between the fourth and the fifth section

The cabasa sample is also used with the intention of associating this section with the previous section. Cabasa sound resembles a filtered noise sound, albeit with percussive envelope. The cabasa is the only instrument keeping a slow and static rhythm in this section. This tempo and the rhythm are exactly the same as SOUND 4-A.

The amplitude of the submix of percussion section is used as the source of side-chaining and modulates SOUND 5-A (pad sound). This is different from the previous section's "timbre side-chaining", because this section uses "normal" side-chaining, which means the amplitude of one part affects the amplitude of another part. Although the effect of side-chaining may not be obvious, I found that it promotes the better blending of pad sound and percussion sounds in the mix.

Towards the closing of the section, percussion instruments except for xylophone are gradually eliminated alongside of the gradual timbral changes of pad sound and xylophone, making the overall impression that pitch becomes more prominent in the end.

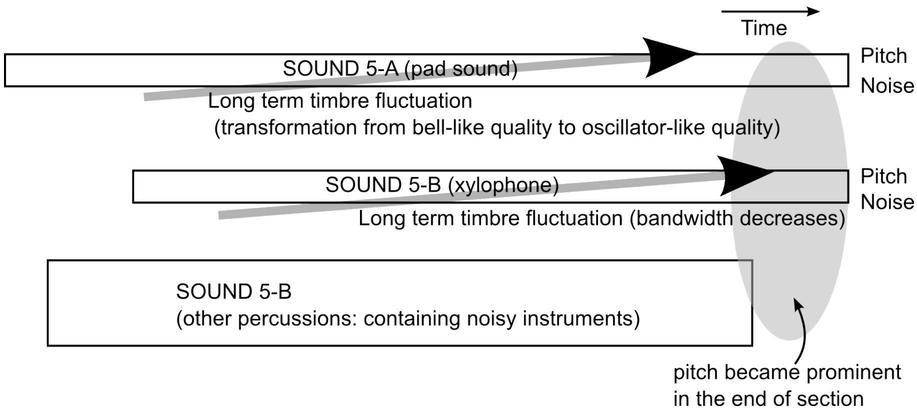


Figure 4-3-11: Long-term timbral change in the fifth section

6. The sixth section (12:42-16:38)

The last section is composed of the following sounds:
SOUND 6-A: Percussive noise made with granular synthesis
SOUND 6-B: Sustained band-limited noise
SOUND 6-C: Phase modulation harmonic sound
SOUND 6-D: Filtered pulse
SOUND 6-E: Low frequency sustained sound

In the end of previous section, the xylophone keeps a 3-2 rhythm, as the tempo is accelerated, the sound level becomes quieter. In the beginning of this section, the xylophone sound suddenly turns into a loud noise (SOUND 6-A). However, noise still keeps the 3-2 rhythm. In other words, timbre and amplitude make a big contrast at the transition of these two sections, and the same rhythmic pattern ties those contrasting sounds together.

Two of my previous pieces (*Flock* and *Colour Composition 2*) focused only on gradual sonic transformation and avoided any abrupt change. The reason I stuck to gradual changes was to help listeners to pay close attention to these sonic complexities. As explained in chapter 3-6, slow music is a valid approach for this purpose. However, I was motivated to introduce more surprises into my music after composing these previous pieces. This transitional part from xylophone to noise is the result of my attempt to bring surprise without loosing musical consistency and an overall attention to timbre. Fixing some parameters (in this case, rhythm) enabled larger change to happen to other parameters (in this case, timbre and amplitude) while maintaining the musical unity.

The percussive noise is made by granular synthesis. The reading speed of buffer, the reading position in the buffer, and the duration of grains are modulated with white noise. The reason of using granular synthesis here is to obtain varieties of different noisy textures by way of alternating granulated samples. This method enables me to make rhythmic sequences of noise, which are characteristically in continuous change. The samples are taken from 6 different field recordings with a high degree of noise, such as the sound of waterfall and firecracker. The peculiarities of original samples and the modulation of parameters with white noise bring noisy quality to resultant sounds.

In the middle of the section, a sustained band-limited noise (SOUND 6-B) supersedes a sequence of rhythmic noise (SOUND 6-A). Figure 4-3-12 shows the patch to make this band-limited noise.

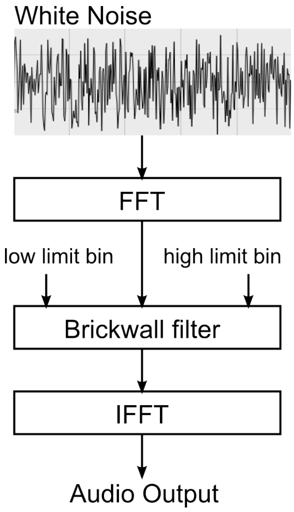


Figure 4-3-12: Signal flow to make band-limited noise in the sixth section

After a while, sequence made with PMOsc (phase modulation sine oscillator pair) comes in. (SOUND 6-C) Each time this harmonic sound hits, the frequency range of band-limited noise (SOUND 6-B) changes by shifting the low and the high limit of band. The pitch of phase modulation sound and the frequency range of noise are related (Figure 4-3-13), thus the phase modulation sound and noise have the relationship both in rhythm and pitch (the shift of noise band might be considered as timbral change instead of pitch change for some listeners). To enhance the unity of the piece, 3/4 steps brownian motion used in the first and the second section is applied again to determine the pitch sequence.

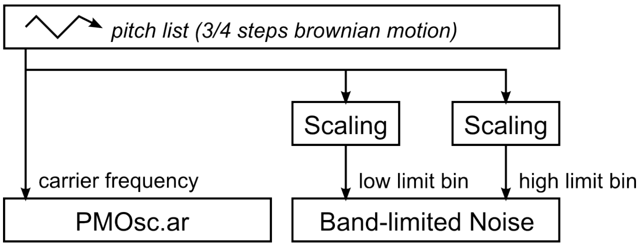


Figure 4-3-13: Frequency relation between phase modulation sound and band-limited noise

The density of PMOsc sound gradually increases and the change of noise accelerates accordingly. The modulation index of PMOsc is changed with brownian motion, too. This change of modulation index gives the continuous timbral fluctuation. It turned out that the combination of density increase and acceleration of timbral changes creates the impression that timbre is getting brighter in the end.

The intention behind the usage of filtered pulse (SOUND 6-D) is to reinforce the connection between the first section and the last section. However, filtered pulses of two sections have a little different sound quality. In the first section, impulse simply goes to band pass filter. In this section, exponential decay is applied to impulse before it goes to band pass filter. The directions of long term timbre movements are also different in two sections. In the first section, the bandwidth of filters applied to pulses gradually narrows in order to increase the quality of pitch. In this section, the opposite timbral movement occurs.

The density of pulse increases towards the end of the piece concurrently with the acceleration of SOUND 6-B and SOUND 6-C. This gesture makes a vertical relationship between pulse and other musical elements.

In the end of the piece, SOUND 6-E (low frequency sustained sound) is played. This synth, with the addition of multiple sine waves, is the same one used for Sound 1-A. In a similar way to filtered pulse, it is intended to enhance the unity by using the same sound in the beginning and at the ending of the piece.

7. The complexity in this piece

The relationships among materials used in this piece are visualized in Figure 4-3-14. Although the number of materials is not a lot for 16 minutes piece, each material is vertically and horizontally connected to the materials as another, accordingly a big network is created. This network is the embodiment of my approach to complexity: making relationships of materials instead of increasing the number of materials. By engaging complexity with a limited number of materials, it becomes possible to maintain listener's sustained attention to inner evolutions of each material. Additionally, this networking approach helps to increase the unity of the piece.

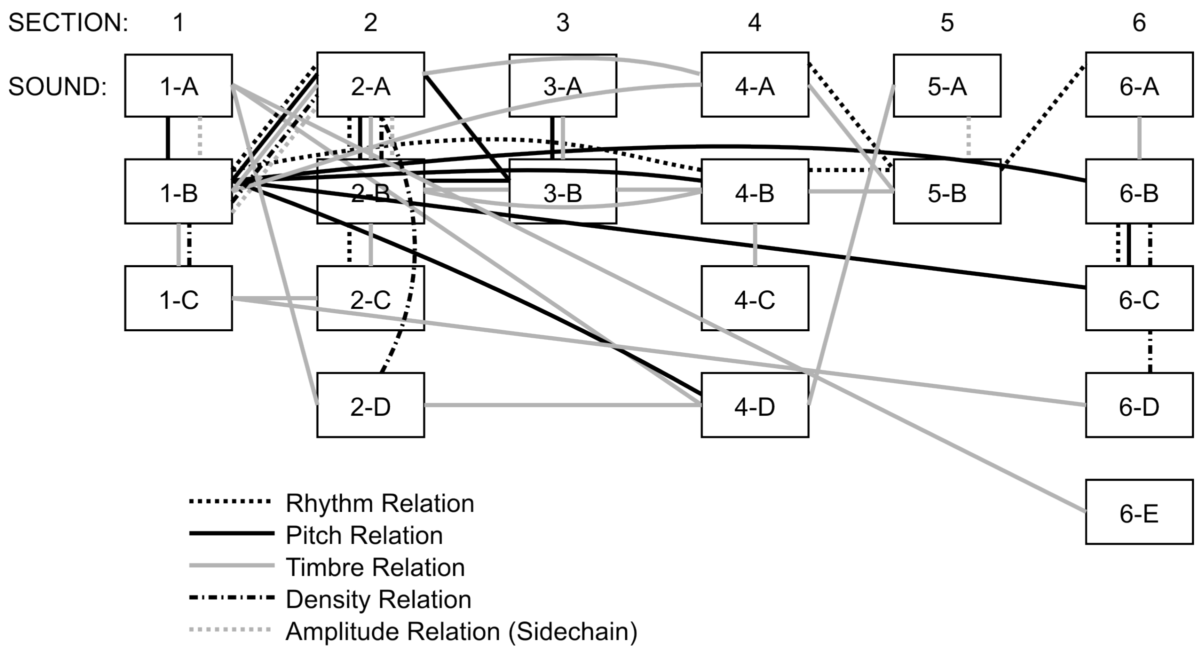


Figure 4-3-14: Network of sound materials used in the piece

Conclusion

The purpose of the present study is to extend the expressiveness of electronic music by exploring timbral movements in composition.

Although the birth of electronic music made it possible to produce timbral movements in sound materials, there has been a difficulty of controlling timbre efficiently with conventional synthesis methods. Band-limited oscillator I suggested is an alternative technique to solve the problem of controllability. By limiting the number of parameters for shaping spectrum, it becomes easy to manipulate timbral movements.

Timbre is a phenomenon that appears as a result of the combination of various physical and musical conditions. Thus, it is almost impossible for composers to control timbral movements at will. Instead, what we can do at best is focusing on one facet of this complex phenomenon by extracting parameters and finding musical behaviors by experimenting with the extracted parameters.

The development of new synthesis techniques can be considered as the extraction of parameters that allow us to obtain new timbral movements, or spectral movements in more technical term. For example, FM synthesis made it possible to change the balance between the amplitudes of the carrier and sideband components by moving parameters. The parameters of Band-Limited Oscillator permit to control the spectral slope in dB scale and the balance between even harmonics and odd harmonics.

One of the main points of this thesis is that timbral movements contribute to the unity and the complexity of music by creating relationships among multiples materials. Newly introduced parameters offer the timbral movements that have not been available before, and then newly obtained timbral movements allow new ways of making relationships among materials.

In sum, the exploration of timbre in the context of electronic music composition is done by the following cycle: developing new technologies, finding new musical gestures with the newly available technologies, and integrating the newly found gestures into compositions. This is the process of discovery, which requires both the scientific researches and artistic practices.

Electronic music is the relatively new field in the long history of music and we only know a part of its potential for now. I believe this heuristic approach for the exploration of timbre let us keep finding undiscovered sonic gestures and musical expressions.

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Appendix 1
Contents of accompanying CD

The contents of CD can also be downloaded from here:
https://dl.dropboxusercontent.com/u/21585549/So_Oishi_Master_Thesis.zip

The CD contains the following 5 folders:

1 Audio Files

contains the following tracks covered in this text:

1 Flock - stereo.aif
2 Flock - granular material.aif
3 Flock - birds material.aif
4 Colour Composition 2 - stereo.aif
5 Colour Composition 2 - excerpt – glissando cloud.aif
6 Colour Composition 2 - excerpt - 3m20s - 5m44s.aif
7 Colour Composition 2 - excerpt - 3m20s - 5m44s - alt ver.aif
8 Colour Composition 3 - stereo.aif

2 SuperCollider UGen BLOsc Pack

contains the source code of SuperCollider UGen BLOsc and files necessary for installing this UGen. See Appendix 2 for the detail of installation.

3 SuperCollider UGen BLOsc2 Pack

contains the source code of SuperCollider UGen BLOsc2 and files necessary for installing this UGen. See Appendix 2 for the detail of installation.

4 SuperCollider Code Examples

contains sample codes demonstrating UGens BLOsc and BLOsc2. These codes are explained in chapter 2-4 and 2-5.

5 Thesis

contains PDF version of this thesis.

Appendix 2
How to install SuperCollider UGens BLOsc and BLOsc2

SuperCollider can be downloaded from SourceForge: http://supercollider.sourceforge.net/

UGens BLOsc and BLOsc2 have been tested in Mac OS X 10.8, 10.9, and 10.10 running SuperCollider 3.6.6.

Installation
In the companion CD, a folder titled "2 SuperCollider UGen BLOsc Pack" contains files necessary to install BLOsc. A folder titled "3 SuperCollider UGen BLOsc2 Pack" contains files for installing BLOsc2.

If SuperCollider is open and the localhost is in use, quit the server.

Move the UGen files with extension .scx and the class definition files with extension .sc to the SuperCollider Extensions folder in the users library for Application Support, e.g. /Users/user/Library/Application Support/SuperCollider/Extensions
If the Extensions folder does not exist, it should be made.

Move the help files with extension .schelp to the Extensions folder in HelpSource/Classes, e.g. /Users/user/Library/Application Support/SuperCollider/Extensions/HelpSource/Classes

Recompile the class library (Language > Recompile Class Library) and reboot the server within SuperCollider IDE.

In case SuperCollider does not deal with new UGens correctly, quit SuperCollider and load it again.

Modifying the software
The C++ codes for the plug-ins with extension .cpp and CMakeLists.txt files are also included. These files can be used to modify UGens.