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APPENDIX I

Guide to the Basic Concepts and Techniques of Spectral Music

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KEY WORDS: Spectral Music; techniques; algorithm; harmony; frequency.

The music discussed in these two issues makes use of many ideas, terms and techniques which may be unfamiliar to many of its readers. I have asked the individual authors not to concentrate on these technical issues in their contributions but, rather, to emphasize the musical and aesthetic ideas being discussed. To provide the context and detail needed to properly introduce and explain this material to readers for whom it is new I have written the text which follows.

This appendix is divided into several major categories, each of which is presented as a succession of major terms, ideas or techniques that build upon one another. A brief perusal of these subjects should clarify the concepts discussed in the articles of these issues, while a closer scrutiny should enable the interested reader to obtain more complete explanations (including, when necessary, the relevant mathematical information). I will be as parsimonious as clarity allows concerning the musical and aesthetic consequences of subjects discussed.

Derivation of pitch aggregates from spectral models

frequencies vs. notes

One of the most basic changes introduced by spectral composers was the generation of harmonic and timbral musical structures based upon frequencial structures. The frequency of a pitched sound is the number of times that its regular pattern of compressions and rarefactions in the air repeat each second. This value is expressed in Hertz (Hz) or cycles per second. Contrary to the linear structure of notes and intervals, where distances are constant in all registers (the semitone between middle C and D-flat is considered identical to the semi-tone between the C and D-flat three octaves higher), the distance between the frequencies within the tempered scale and the potential for pitch discernment of the human perceptual apparatus is neither linear nor constant: it changes in a way that is completely dependent upon register. Viewing structures from the perspective of frequencies gives access to a clear understanding of many sounds (like the harmonic spectrum) whose interval structure is complex, but whose frequency structure is simple. It is also extremely useful for creating sounds with a high degree of sonic fusion, since the ear depends on frequency relations for the separation of different pitches. Further, a frequency-based conception of harmonic and timbral constructions allows composers to make use of much of the research in acoustics and psychoacoustics, which look into the structure and perception of natural (environmental) and instrumental sounds, providing models for the way in which various frequencies are created and interact to form our auditory impressions.

the equal-tempered scale (from the perspective of frequencies)

The equal-tempered scale is based on the division of an octave into a number of logarithmically equal parts (not linearly equal) — in the case of the chromatic scale this is 12 parts. An octave is defined as the distance between a note and the note with twice its frequency (thus if the frequency of A4 is 440 Hz, the frequency of the note an octave higher, A5, is 2 * 440 or 880 Hz, and the frequency of the note an octave lower, A3, is 440/2 or 220 Hz). Microtonal scales follow the same principle, but divide the octave into different numbers of logarithmically equal steps (24 of them for the quarter-tone scale, 48 for eighth-tones, etc.). The formula for calculating the chromatic scale is the following: frequency of note x + one half-step = frequency of note x times two to the power of one over the number of steps in the octave (1/12 for the chromatic scale). To calculate notes in the quarter-tone scale the only change necessary is to use two to the power of one twenty-fourth and thus the equation is: frequency of note x + one quarter-tone = frequency of note x times two to the power of one twenty-fourth. To calculate an actual scale you begin with the diapason (the reference pitch, for example A4 = 440 Hz), then calculate the notes above and below it (to calculate down instead of up, you divide instead of multiply). Table 1

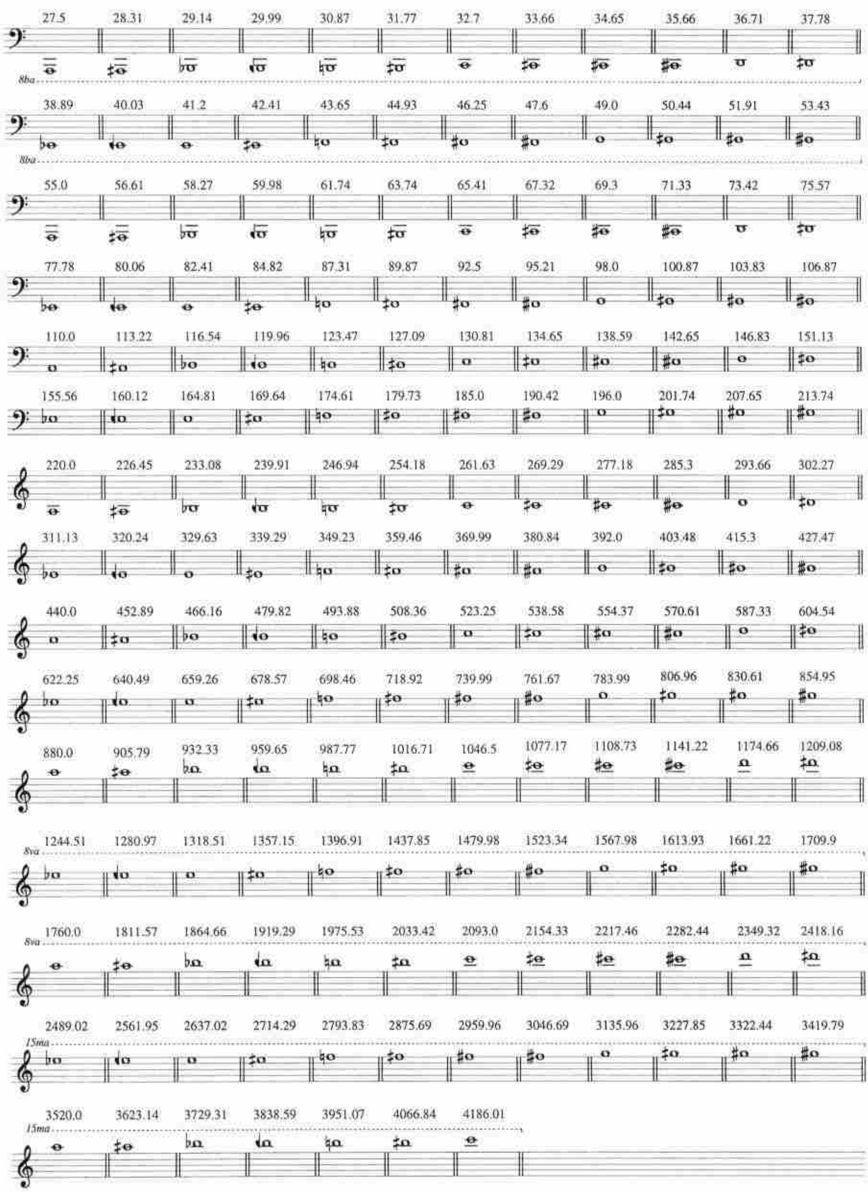


Table 1

shows the frequencies corresponding to all the notes in the usual range of musical compositions of a tempered quarter-tone scale with a diapason of 440 Hz.

microtones

For spectral composers, microtones are not the result of scales built on frequency ratios, nor even one of tuning. Instead, the microtones in spectral music are simply approximations of a set of frequencies to the nearest available musical pitches. In most cases, quarter-tones are used for instrumental music (with some eighth-tones in very slow tempos and occasional revertions to semitones in very fast tempos or for keyboard instruments). This approximation is often a last step, allowing the musical structure to be generated in its most precise form (frequencies), then approximated to the nearest available pitch depending on the details of the instrumental abilities and context. This also allows many spectral composers to tailor difficulty to individual realizations, adding or removing difficult notes in a way that does not change the underlying structure, but merely refines or coarsens the approximation of the abstract musical structure. Since the ear analyses structures based upon their frequency structure, the ear is able to hear past these approximations and hear the underlying frequency structure whenever the approximation is within tolerable limits.

the concept of additive synthesis

The remaining topics in this section and those of the second section will deal with frequency structures. Some are abstract, like the harmonic series, some are based upon the analysis of natural sounds, and some are extrapolated from mathematical models of sound. In all cases, this way of looking at sounds comes to us though analytical techniques such as the Fourier Transform (see below) and has its clearest, most intuitive expression in the electro-acoustic technique of additive synthesis. In this technique, the simplest possible sonic components are used: sine waves. Sine waves are described as the simplest sonic components because they are the only periodic wave-forms whose spectra contain only the frequencies of their oscillation (for complete mathematical rigor, they would have to be of infinite duration for this to be absolutely true — in the mathematical sense — it is nonetheless essentially true at all time-scales longer than one period). This property of sine waves makes them both an ideal medium into which to decompose sounds and an ideal unit from which to build them. Fourier's Theorem states that any periodic sound can be decomposed into a number of sine waves (in some cases, however, this may not be a finite quantity) and also provides the corollary that the combination of these elementary units can rebuild the original sound. The technique of additive synthesis applies this principle, building up complex sounds through the combination of a large number of elementary ones (sine waves). This technique is extremely powerful, in principle, since any sound can theoretically be synthesized in this way. In practice, however, it is not that simple. The number of components needed for a convincing recreation of a given sound (resynthesis) is often enormous and the amount of data needed to generate convincing sounds was often too much to handle before the existence of automatic analysis/resynthesis tools. Actual sounds resulting from additive synthesis are often too simple and static to create satisfying musical sounds, since all changes must be explicitly defined (whereas certain other techniques generate spectral flux automatically). The great and enduring advantage of this technique is conceptual. It provides the clearest, most intuitive way for us to conceive of hearing and creating sounds. By listening closely to any sound, it becomes possible to hear the separate components, and, by adding sounds together it is easy to hear the global sound color, or timbre emerge and evolve.

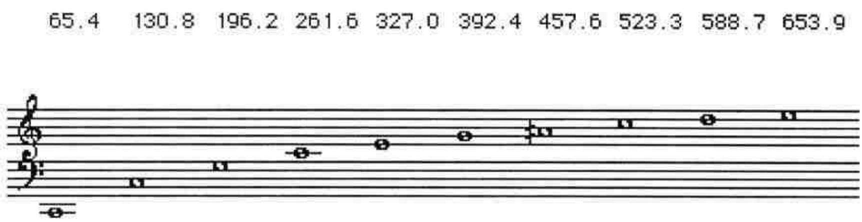
instrumental or orchestral synthesis

Perhaps the most important idea emerging from early spectral music (though it was presaged in other musics) was the idea of instrumental (or orchestral) synthesis. Taking the concept of additive synthesis, the building up of complex sounds from elementary ones, and using it metaphorically as a basis for creating instrumental sound colors (timbres), spectral composers opened up a new approach to composition, harmony and orchestration. The sound complexes built this way are fundamentally different from the models on which they are based, since each component is played by an instrument with its own complex spectrum. Thus the result is not the original model, but a new, much more complex structure inspired by that model. The sounds created in this way keep something of the coherence and quality that comes from the model while adding numerous dimensions of instrumental and timbral richness and variety. All of the frequency structures described below and the procedures for approximating frequencies described above can be combined to transform any of these structures into models for orchestrally synthesized timbres. The potential use of the same model for generating synthetic sounds (through additive synthesis) and orchestral ones (through instrumental synthesis) is also a reason why mixed electronic and acoustic music has played such an important role in the output of spectral composers.

the harmonic (or overtone) series

The harmonic (or overtone) series is a mathematical phenomenon of sound which was recognized at least as early as the Greeks. It is defined by an integer relation between a fundamental frequency and the other components of a sound. In the earliest studies of the phenomenon it was found by generating harmonics on a monochord and later Helmholtz was able to show (through sympathetic vibrations) that this structure exists within the sounds of instrumental pitched notes — see below. The equation for generating harmonic partials from a fundamental frequency is: frequency = rank * fundamental; where ranking is an integer defining the partial number and both fundamental & frequency are expressed in cycles per second (Hz). For example, if the fundamental is 440 Hz then the second partial is 2 * 440 or 880 Hz, the third partial is 3 * 440 or 1320 Hz, the fourth is 4 * 440 or 1760 Hz, etc. To equate these frequencies to musical pitches they must be approximated to the finest available resolution (see above).

The first ten partials of a harmonic series (notes approximated to the quarter-tone, frequencies marked above):



This and all following musical notation examples were created in the program PatchWork and are used with the permission of IRCAM.

harmonic spectra

Pitched sounds are often formed by combinations of sonic components (partials) which belong to a single harmonic series whose fundamental is heard as the pitch. The relative amplitudes of these various partials — at a given moment and as they change in time — determine the color or timbre of the sound. The simplest harmonic spectrum is a sine wave with only the fundamental and no other partials. Most sounds, however, contain many partials of the fundamental, and the fundamental pitch can be deduced by the ear even when it is not explicitly a member of the spectrum.

instrumental spectra

Western instruments have been developed, for the most part, to have spectra which are very close to pure harmonic spectra, so as to emphasize clarity of sound and pitch. However, because of the physical system of sound production that they use, the sounds are never completely harmonic. There is often a noise component, caused, for example, by the breath of a wind player or the scraping of the bow on the string, and the harmonic part of the spectrum is often very slightly stretched or compressed by the physical properties of the vibrating medium (an example of this is the slight stretching of the harmonic partials in the piano which forces tuners to make slightly 'large' octaves in the upper register). Other important characteristics which are critical in determining the timbre of instrumental spectra, and vocal spectra for that matter, are the relative amplitudes (and presence or absence) of various partials, their formants, their envelopes and their attack transients — see below.

relative amplitudes of partials within instrumental spectra

One of the defining aspects of instrumental timbre and of the characteristics of different registers on the same instrument is the relative amplitudes of the various partials. For example, the flute's weak fundamental tone (in comparison in its to its second partial) in its low register diminishes the focus of its pitch and makes the notes in this register difficult to hear at a distance and causes them to be easily masked by other instruments. There are many other characteristic examples: the clarinet spectrum tends to emphasize only odd numbered partials; fortissimo brass instruments tend to have dissonant upper partials (the seventh or the ninth) as the loudest components of their spectrum — this is the cause of a 'brassy' sound; the higher portion of an instrument's range tends to have fewer partials than the lower portion does — this makes the sounds more penetrating and concentrated, but can also make them shrill and harder to artistically modify their timbre; etc. A major influence on the relative amplitudes of these partials are the formants caused by the instrument's physical resonance system.

formants

When physical bodies vibrate, they act, to a certain degree as filters, emphasizing certain bands of frequencies and attenuating others. For most instruments this is a fixed part of their construction. This is why particular frequency regions will be highlighted or masked regardless of the pitch played (since these regions are defined by frequency, they will affect the upper partials of low notes and lower partials of higher notes, but they will always be centered around the same frequencies). In an instrumental context, formants are one of the main clues that allow us to hear that the high notes and low notes of an instrument come from the same source. Without them, we would be tempted to hear the chalumeau (low) and clarino (upper-middle) registers of the clarinet as the different instruments which gave their names to these registers and not as different inflections of the same instrument, as we do. The human voice makes especially important use of formants, since muscles are capable of changing the physical parameters of the resonating body (the throat, mouth and nasal cavities) and thus creating different types of formants. This is, in fact, the mechanism which allows us to hear vowels. Each vowel is defined by its characteristic formants and can be produced on any pitch. For example, a soprano singing the vowel 'e' has formants at 350, 2000, 2800, 3600 and 4950 Hz, one singing the vowel 'a' has formants at 800, 1150, 2900, 3900, 4950 Hz, etc. (The reason that it is often difficult to understand a high soprano is that the high fundamental is above the first of the formants and the large space between partials allows the presence of formant regions in which no partial falls, so that no sound can be emphasized by the filtering mechanism. This lack of formant cues prevents us from understanding the vowel sound and therefore, the text.) It has been speculated that the intricacy of timbral hearing required to discern the formants of speech, for anyone who speaks and understands an aural language, are part of the reason that timbral hearing is much more widely and finely developed than the kind of pitch and interval hearing taught in Western music (it is easier for untrained listeners to differentiate a flute and an oboe playing the same pitch after a brief demonstration of the two sounds then it is for them to differentiate a major and minor third after the same kind of presentation, although the latter difference is quantitatively very much the greater).

example

The following examples shows the spectra of the sustained portion of a flute sound playing a note mezzo-forte in two different registers:

d4 :

Partial	frequency	amplitude	
1	23.665	.074692	
2	**587.330**	**.210554**	— second partial three times louder than fundamental
3	880.995	.052184	
4	1174.660	.020801	
5	1468.325	.013828	
6	1761.990	.002447	
7	**2055.655**	**.007147**	— formant a (three times louder than surrounding partials)
8	2349.320 .	003839	
9	2642.985	.000181	
10	2936.650	.000136	
11	3230.315	.000419	

12	**3523.980**	**.001390**	— formant b (three times louder than surrounding partials)
13	3817.645	.000129	
14	4111.310	.000186	
15	**4404.975**	**.000297**	— formant c (two times louder than surrounding partials)
16	4698.640	.000108	
17	4992.305	.000114	
18	5285.970	.000182	
19	5579.635	.000192	
20	5873.300	.000077	
21	6166.965	.000067	
22	6460.630	.000060	
23	6754.295	.000058	

d6 :

Partial	frequency	amplitude	
1	1174.659	.304999	— **no partial at 2000 Hz for formant a**
2	2349.318	.007111	
3	**3523.977**	**.010688**	— formant b (significantly louder than surrounding partials)
4	4698.636	.000467	— **no partial at 4400 Hz for formant c**
5	5873.295	.001536	
6	7047.955	.000183	
7	**8222.613**	**.000259**	— formant d (two times louder than surrounding partials)
8	9397.272	.000165	
9	10571.932	.000104	
10	11746.591	.000075	
11	12921.250	.000075	

This example illustrates the relative amplitudes of the partials and how they change with register. It also demonstrates the constancy of some formants, illustrating their role in preserving the identity of the flute.

spectral envelopes

The previous subjects have addressed the internal amplitude relationships of the partials within a spectrum and the overall amplitude of the sound as if they were static. This is, of course, not true. The changes in the overall amplitude as well as the relative changes of the partials are critical to creating our impression of timbre. However, for the most part this will be dealt with below, in the section on dynamic FFTs. These changes are often complex and difficult to model and thus have only come to play a critical role through the growing availability of tools for dynamic analysis. As concerns the type of abstract modeling which we are treating in this section, only a few basic principles will be detailed (this follows the historical evolution, wherein early spectral music treated envelopes in a fairly schematic manner, but newer pieces make use of much more sophisticated treatments). The most basic aspect of an envelope is its global amplitude motion. A percussive sound, like a drum, has a very fast attack (the portion of the envelope where the amplitude is increasing) a short or non-existent sustain (the portion of the envelope where the sound is essentially stable) and a decay (the portion of the envelope where the sound is fading away) which will be longer or shorter depending on the amount of resonance of the instrument. A more subtle, but also important, aspect of timbre is the spectral envelope, which determines the appearance, disappearance and changing relative amplitudes of various partials. For example metallic percussion sounds begin with many partials, but the upper partials decay quickly leaving fewer and fewer partials in the later resonance. Another example is brass sounds which begin with few partials, then, as the attack progresses, higher and higher partials enter; during the decay this procedure reverses as the higher partials leave first and finally only the lower partials are present at the end. Also an important part of spectral envelopes is spectral flux. This is the amount of variation within a sound during its evolution and even its sustained portion. Although this change often seems random, it is essential to our hearing a sound as 'natural.' The failure to mimick this flux, due to the difficulties it poses, is often the reason that many electronic sounds are instantly identifiable as 'artificial.' While especially significant for sound synthesis, these phenomena have also led many spectral composers to introduce micro-variations into their instrumental timbres, mimicking this attribute of natural sounds.

attack transients

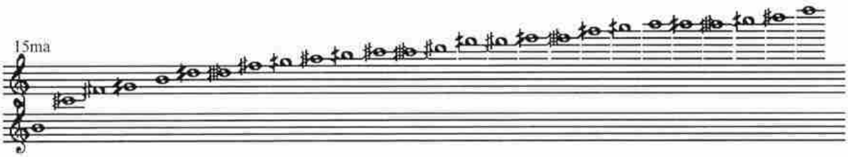
Another extremely important aspect of instrumental timbre which is temporally unstable is the attack transient. This transient is a coloring of the spectrum which is present only in the first part of the sound. It is generally noise-like and is often caused by mechanical parasites in the physical production of the sound (for example, the scraping of the bow before the pitch has stabilized in a string instrument, the impact of the hammer in a piano, etc.). This portion of the sound, while difficult to analyze, is extremely important to the perception of timbre. (It has been shown that if the attack is removed it becomes very hard to identify instrumental timbres correctly.) Fortunately, for modeling purposes, the exact frequency content of the attack transient seems to be less important than its presence. The amplitude of the transient relative to the harmonic portion of the spectrum and its envelope seem to be central to perception. Therefore, while many spectral composers have worked with the idea of attack transients and have sought to include and manipulate them in both electronic and instrumental synthesis, this modeling is rarely based on precise models emerging from analyses and is, instead, more intuitively and metaphorically based on the concept.

non-harmonic spectra

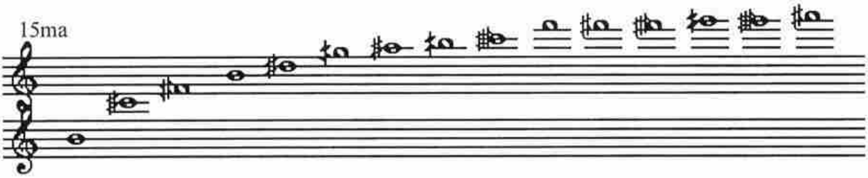
As was stated above, most pitched instruments have spectra which are very close to harmonic. However, many instrumental sounds with less defined pitch, or no identifiable pitch at all, have spectra which are non-harmonic. There are an enormous, if not infinite, number of possible non-harmonic spectra found in physical instruments (artificial non-harmonic spectra will be dealt with below). They can, however, be grouped into a few classes of spectra. The first category is colored noise, like a guiro or maracas or blowing through a flute. This type of sound can be modeled as a resonant or band-pass filter applied to white noise (sound with equal amounts of energy in all frequencies). Spectra of this type are also found when metal plates are struck (although in this case the source is an impulse rather than sustained white noise). Another type of non-harmonic spectrum is found in instrumental multiphonics or bells. These sounds have multiple superposed harmonic spectra sounding simultaneously and, in some cases, producing beats between them. Other non-harmonic spectra are sounds in which the small amounts of spectral stretching or compression, mentioned above, have been pushed to the point where they interfere with the ear's perception of a fused, pitch producing, timbre. In all these cases, non-harmonic spectra are fundamentally different from harmonic spectra in that they do not produce the same clear sense of spectral fusion (the cognitive property which allows us to hear the components of a sound collectively, perceiving the different components only as they influence the timbre of the global sound and not as individual pitches) or well-defined pitch (without spectral fusion the different components are heard separately, giving conflicting pitch cues). While this class of spectra is certainly richer than that of harmonic spectra, it is often difficult for listeners to make the kinds of fine distinctions with these sounds that they can with the more familiar harmonic sounds (this is certainly a cultural phenomenon since Balinese listeners, for example, distinguish much more easily between different metallic percussion sounds than do Westerners, who are less familiar with these sounds).

example

The non-harmonic spectrum of a cow-bell (shown approximated to the nearest quarter-tone) has a large number of weak (low amplitude) upper partials when first struck:



The highest of these partials disappear almost immediately, leaving a spectrum of medium complexity:



The upper partials of this spectrum continue to fade, leaving only the most significant partials to resonate:

Note that due to the effects of beating and modulation between partials, some notes change pitch slightly during the sonic evolution.

artificial (harmonic) spectra

The relations described above concerning the harmonic series and instrumental spectra all lend themselves to simple modeling. From the very beginning of the spectral movement, composers have taken the simple mathematical expression of the harmonic series described above (frequency = rank * fundamental; where ranking is an integer defining the partial number; the fundamental and the frequency are expressed in Hz) and used it to create abstract spectra. Rather than analyzing the spectrum of a given sound, they created sounds that were harmonic, but had not previously existed. Any combination of harmonic partials built upon the same fundamental share certain acoustic properties, to which the ear is sensitive, and create a high degree of fusion. By using novel combinations of partials and amplitudes spectral composers have able to create new, artificial sounds that keep much of the naturalness of acoustic sounds and that give spectral music the kind of sonorous resonance that has often been remarked in this music.

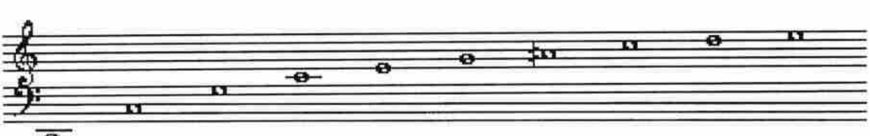
distortions of harmonic spectra

As was mentioned above, many instrumental spectra are not perfectly harmonic. They are slightly stretched or compressed. To model this effect (and also to extend it to extreme distortions that, while not found in nature, offer interesting musical possibilities often used by spectral composers) an exponent is added to the equation for harmonic partials, producing the following equation: frequency = rank * fundamental raised to the x power; where ranking is an integer defining the partial number; the fundamental and the frequency are expressed in Hz, and x is a value greater than zero. If the exponent x is less than one the spectrum is compressed, greater than one, it is stretched; and equal to one, it is harmonic. In nature, these values will always be quite close to one (e.g. 0.98 or 1.03, etc.), but for musical purposes they can be much more varied.

example

The first ten partials of a harmonic spectrum (notes approximated to the quarter-tone, frequencies marked above):

65.4 130.8 196.2 261.6 327.0 392.4 457.6 523.3 588.7 653.9

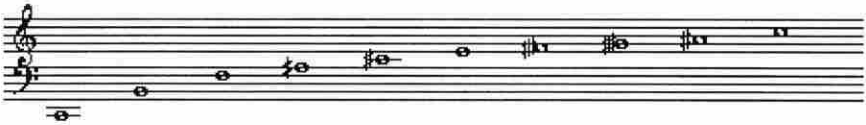


Stretched version of the same spectrum (distortion coefficient of 1.1):

65.4 140.2 219.0 300.5 384.1 469.4 556.1 644.1 733.2 823.3

Compressed version of the same spectrum (distortion coefficient of 0.9):

65.4 122.0 175.8 227.7 278.4 328.0 376.8 425.0 472.5 519.5



Another frequently used distortion technique (which also compresses or stretches the spectra) is to change the highest and/or lowest notes of the spectrum and then to re-scale the interior notes of the aggregate in such a way as to preserve the relative spacing between the frequencies.

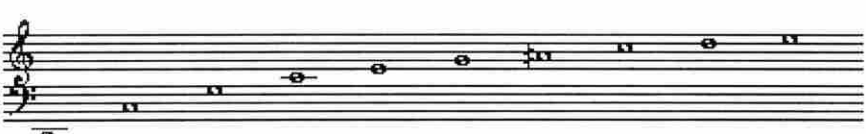
frequency shifted spectra

A different type of spectral distortion, inspired by analog electronic devices not by nature, is frequency shifting. This technique, instead of creating a distortion which gets ever larger as the frequencies increase (since it is exponential), adds a constant value (in Hz) to the frequencies of all the partials. Therefore the equation for partials in a frequency shifted spectrum is frequency = rank * fundamental + shift-value where ranking is an integer defining the partial number; the fundamental, frequency and shift-value are expressed in Hz. The effect of this distortion becomes progressively less significant as frequencies increase, since the percentage of distortion decreases. This treatment has a very characteristic sound which is quite distinct from other types of distortions.

example

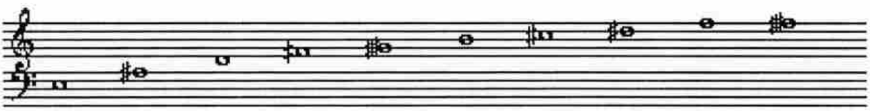
The first ten partials of a harmonic spectrum (notes approximated to the quarter-tone, frequencies marked above):

65.4 130.8 196.2 261.6 327.0 392.4 457.6 523.3 588.7 653.9



The same spectrum frequency shifted by 100 Hz :

165.4 230.9 296.4 361.5 427.0 492.2 557.6 623.0 688.8 753.8



The same spectrum frequency shifted by –10 Hz :

55.4 120.8 186.3 251.5 316.9 382.6 447.7 513.1 578.6 644.2

modulations (general)

The spectra discussed up to this point are based on what physicists refer to as simple oscillations. By this they mean that there is only one periodic element to the wave-form, though that element may be complex. However, in many situations, one sound interacts with a second independent sound. The most familiar form of interaction is for one sound to modulate the other. Three types of modulation which have frequently been used by spectral composers will be detailed below: amplitude, frequency and ring.

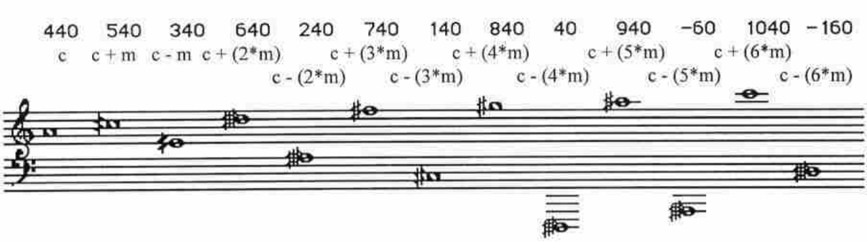
amplitude modulation

This type of modulation, which is used for AM radio, is most familiar in music as amplitude vibrato, such as one finds in flute playing. Amplitude modulation is also important for its role in creating spectral flux (described above) in instrumental timbres. Most partials in nature have amplitudes which are constantly varying, even when the general impression is of a constant level. This aspect is modeled in many synthesis applications with jitters (random envelope generators) or low frequency oscillators (less then 20 Hz) modulating the main amplitude generator. Auditory rate amplitude modulation (modulations faster than 20 Hz), which generates new audible partials which are distinct from those belonging to the two interacting sounds, have not really been used much either in electronic or spectral music.

frequency modulation

Frequency modulation (FM) is the most prominently used modulation for musical applications. In its simplest form, frequency modulation can take the form of pitch vibrato, like that used by string instruments. In the seventies, John Chowning developed the technique of auditory rate FM, which modulates a sine wave carrier by a sinusoidal modulator with a frequency faster than 20 Hz. This type of modulation creates side bands (partials on either 'side' — symetrically above and below — of the carrier frequency) in the generated spectrum. These side bands have the advantage of exhibiting a great deal of spectral flux as the modulation depth, or index, changes. (Modulation depth and index are ways of expressing the amount of modulation that effects the carrier; the greater the modulation depth, or index the more side bands that are present in the modulated spectrum. The relative amplitudes of the side bands are also dependent upon this depth or index.) While these changes are quite different from those found in instrumental spectra, Chowning demonstrated that in many cases the existence of fluctuations is more important than the exact structure of those changes. Therefore, this technique could generate relatively satisfying sounds with only two oscillators, whereas additive techniques, for example, might require dozens or more. This efficiency led Yamaha to license the technique for their synthesizers, starting with the DX series. In the same way that spectral composers were modeling and analyzing instrumental sounds for the creation of orchestrally synthesized timbres, they also looked to the FM technique. The spectrum produced with this technique is expressed with the following equation: frequency = carrier + and – (index * modulator); where index equals 0, then 1, then 2, etc. until the maximum index value has been reached. The amplitudes of frequency modulated spectra follow relatively complex functions, which are often left out of simple computational models. Spectral composers have used this calculational model to integrate electronic FM sounds with instrumental timbres and to create a new category of spectral models for use in all types of pieces.

The following example shows a modulation with a carrier ('c') of A, 440 Hz, and a modulator ('m') of 100 Hz — slightly above a G2 (notes approximated to the quarter-tone, frequencies marked above):

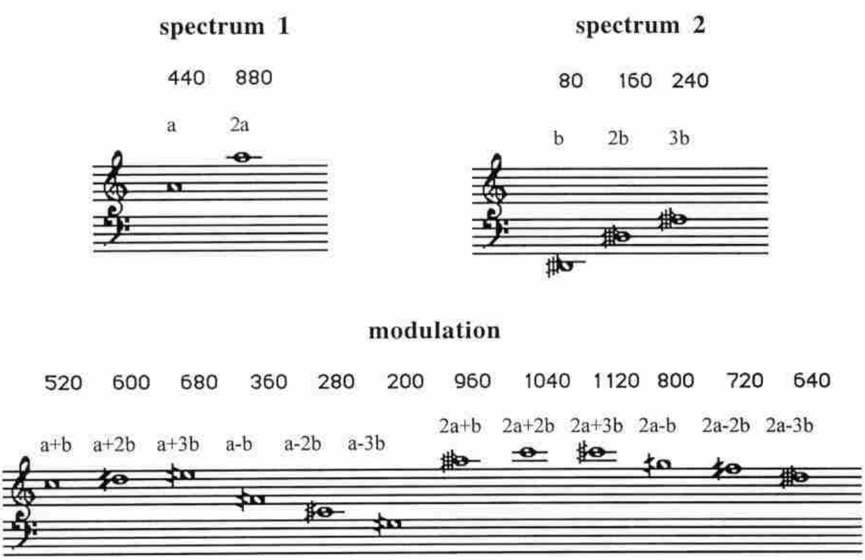


Note that negative frequencies are heard identically to positive ones, except that the phase is inverted.

ring modulation

Originally an analog electro-acoustic treatment, ring modulators modify complex sounds. In the original implementations, a sound captured by a microphone was modulated by a sine wave generator (this is used, for example, in Stockhausen's works *Mixtur* and *Mantra*). The major difference from FM modulation is that this type of modulation is not hierarchic: there is not a carrier and a modulator which modifies it, but two equal sounds both of which are directly present in the resultant sound and both of which are modulated by the other. The spectrum resulting from a ring modulation can be simulated when the frequency of each note of the first spectrum is combined through both addition and subtraction with the frequency of each note of the second spectrum, producing all the possible additive and subtractive combinations of the partials. When the two spectra contain many partials, enormous numbers of combination tones are produced and if the spectra are non-harmonic and rich, the resulting modulation can quickly turn into noise. The number of partials generated will be two times the number of partials in the first spectrum multiplied by the number of partials in the second spectrum.

The following example shows a modulation of a first spectrum built on A, 440 Hz, with the first two partials and a second spectrum built on D 3/4 sharp 2, 80 Hz, with the first three partials, (notes approximated to the quarter-tone, frequencies marked above):



virtual fundamentals

In the above section on the overtone series, the fundamental was explained essentially as the greatest common denominator of a harmonic spectrum. For a distorted, shifted, non-harmonic, or modulation-based spectrum, however, the ear still tends to find a fundamental. (Note: the ear does this even in some instrumental spectra, such as low piano sounds, where the perceived fundamental pitch is absent from the spectrum — the strings being too short to actually vibrate at those frequencies.) Many psychoacoustic algorithms have been proposed which attempt to model this effect by which the ear creates a sort of 'virtual' fundamental in spectra lacking a real one. These algorithms depend on the tolerance of the ear discussed in the above section on approximation. With slight variations, they calculate the greatest common denominator to a given tolerance (which is often user specified). The virtual fundamental has often been used by spectral composers as an ad-hoc measure of harmonicity (lack of tension) or inharmonicity (presence of tension), equating higher virtual fundamentals with greater harmonicity or less inharmonicity and lower ones with less harmonicity, or greater inharmonicity. The motivation for this is that harmonic spectra start with a real fundamental and as they are distorted the virtual fundamental moves in various directions; when these distortions become more and more noise-like, the virtual fundamental descends until, for white noise, the virtual fundamental approaches zero Hz. This technique, however, only works within constrained contexts, especially as regards register. It is often necessary to transpose aggregates to a common register prior to comparing their virtual fundamentals, because a lower register will automatically result in a lower virtual fundamental for the same harmony.

spectra as harmony/timbres

Building global orchestral sounds upon the same models that constitute single instrumental or artificial sounds — spectra — (through instrumental synthesis, described above) creates a powerful ambiguity between the notions of harmony and timbre. Once the instrumental mass is perceived globally, as a color or texture, the notion of harmony becomes less relevant than that of color or timbre. Where the perception is less that of a fused mass, however, it is clear that the notes of these spectral models have taken on a harmonic role. What truly emerges, in fact, is that in spectral music the line between these two concepts has blurred, practically to the point of non-existence. The aggregates in this music are used simultaneously to control the harmonic movement and the timbral evolution. Further, these two types of motion are often indistinguishable. Therefore, in spectral music, at least, it is often more relevant to combine the two concepts into the more general concept of a harmony/timbre; this hybrid concept preserves aspects of both of its component ideas and captures the interdependence and indivisibility that has developed between them.

spectra as reservoirs

Besides building harmony/timbres from the acoustically-based models provided by spectra, many spectral composers use these models as reservoirs. They sometimes treat these reservoirs as modes, from which lines and harmonies can be constructed: the power of this system comes from the fact that acoustic models can generate very large numbers of frequencies (and through approximation, pitches) which can be combined with each other while still guaranteeing an overall coherence. This allows a single underlying structural entity and color to create a proliferation of surface manifestations which are coherently related. Other times these reservoirs are used to provide a metaphor for the musical evolution they are trying to create; for example, moving to higher and higher partials of the same spectrum as musical developments and other parameters augment the ambient tension. Treating spectra as reservoirs and their treatment as harmony/timbres, described in the preceding section, are not contradictory, but complementary. The reservoir approach is often used to provide surface activity within a slower harmonic rhythm and the harmony/timbre approach lends itself to more harmonic passages; however, many different configurations can be found in the music.

Derivation of Pitch Aggregates from Spectral Analysis

spectral analysis — Fourier Transforms

Much of the information already discussed results from information gleaned from spectral analyses, but the actual mechanisms of these analyses has not been explained. They are based on the work of French mathematician Jean Baptiste Joseph Fourier (1768–1830). Fourier showed that any periodic waveform could be decomposed into the sum of a series of sine-waves whose frequencies are at integer multiples of a fundamental (though not necessarily a finite series) with different amplitudes and phases; in other words, all periodic waveforms can be transformed into some type of harmonic series. This is called a Fourier Transform, since the periodic function is transformed into an equivalent Fourier series. While, in theory, the periodic function must be infinite, in practice, several periods of stability are enough for an accurate, though not perfect (in the sense of being able to reconstruct an exactly identical waveform) analysis. Also while the technique in its pure form can create only harmonic spectra, the use of extremely low 'pseudo-fundamentals' allows a good sampling of the spectral energy throughout the auditory range — providing a close approximation of even very non-harmonic sounds. This technique is not well suited, however, for the type of discrete numerical analysis required for musical applications, where infinities cannot be part of the input or results and where a certain amount of quantification error is acceptable. The solution for most applied uses of spectral analysis was the development of the discrete Fourier Transform (DFT). This technique essentially samples discrete time positions of the input signal or function and truncates the Fourier series after a certain number of terms; when this sampling is sufficiently dense, a good approximation of the continuous function is created. This technique was extremely calculation-intensive even for computers, until a class of extremely efficient algorithms for calculating these DFTs was developed. These algorithms depend on a factorable number of points which allows the calculations to be broken into separate parts and reordered in ways that dramatically reduce the number of calculations. They are referred to as Fast Fourier Transforms (FFTs).

Fast Fourier Transform (FFT)

This efficient version of the discrete Fourier Transform is at the heart of all spectral analyses on computers. To perform this calculation on an audio signal, a window of sound must be selected for analysis. This window cannot simply be cut out from the sound without creating major artifacts (frequencies which are created by the mechanism of the analysis rather than as a result of their presence in the sound), but must be extracted with an envelope that creates minimal distortion. The longer the window (temporally speaking), the greater the frequency resolution of the analysis; temporal resolution, conversely, decreases with window length. This is due to the fact that all the sounds within the window are assumed to be unchanging and averaged throughout the window. This creates a situation which is similar to photos (fast shutter times are needed to capture moving objects, slow ones for low light) where compromises must always be made. When studying spectral analyses it is important to remember the significant effect that these parametric decisions have on the final result. In essence, there is not one analysis which represents the reality of an acoustic signal, but many possible analyses which accurately render certain aspects of the sound while distorting others (this is reminiscent of the effect created by different two-dimensional projections of the map of the globe). When selecting these parameters, it is very important to bear this in mind and perform the analyses in a manner well-suited to revealing the aspects that one needs to see. It may even be useful to perform multiple analyses of the same sound.

dynamic FFTs

The above description of FFTs deals with single windows, in which all sounds are averaged together. In order to see the changes within a sound over time, a series of FFTs with windows which advance in time is necessary. Devices like phase vocoders are able to analyze a sound with this technique of FFTs with overlapping windows which advance in time and create a representation of the sound as it evolves. Under optimal conditions, this representation is sufficiently accurate that it can perfectly recreate the sound. While this technique of dynamic FFTs is capable of analyzing almost any sound, a difficulty often resides in assimilating the masses of data that are generated (a set of FFTs produces thousands of amplitudes and frequencies hundreds of times for each second of sound). One of the most familiar and successful means of making this data comprehensible is to generate graphic representations, like sonograms.

sonogram

Sonograms are a graphical representation of the three dimensions of time, frequency and amplitude in two dimensions, with the third dimension of amplitude represented by a clever graphical use of darkness. The advantage of doing this is that the multitude of frequencies with little or no amplitude simply disappear, making the image less cluttered. The program AudioSculpt, created at IRCAM, has provided easy access to sonograms and has had a great influence in making the subtlety of temporally sensitive analysis accessible and manageable. In the past, simplified non-dynamic models, of the sort described in the first section of this paper, were all that composers could easily manage. This sort of representation makes the use of much more sophisticated models possible.

Sonogram of a note played on the harp (see figure 1): the smaller top rectangle shows amplitude on the y-axis and time on the x-axis, the larger lower rectangle shows time (in seconds) on the x-axis and frequency (in Hz) on the y-axis; in this representation, the amplitude is shown by the darkness (in gray-scale).

This type of representation is powerful since it gives a clear and understandable overview of an enormous amount of data.

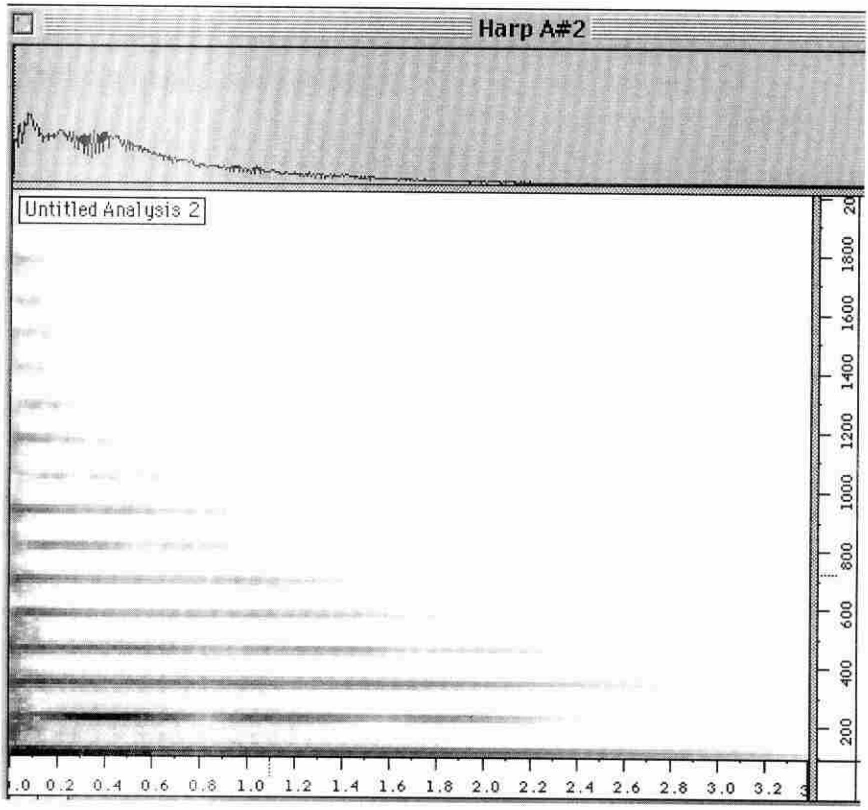


Figure 1 This sonogram was produced with the program AudioSculpt and is used with the permission of IRCAM

data reduction

Composers and scientists have not simply relied on graphical tricks to reduce the amount of data generated by dynamic FFTs; they needed methods for sorting this data and extracting the precise elements needed for different applications. It would be beyond the scope of this article to describe all of these; however two of them that have often been used by spectral composers need to be mentioned.

psychoacoustic algorithms

The most widely used strategy for reducing the amount of data produced by FFTs is to attempt to select the partials of the analyzed sound which are the most important to the perception (or salient). The earliest and most simple strategy, and one used in many spectral pieces, is to select the loudest partials (called the peaks). In the eighties, a desire not just to ground the selection in the quantitative parameter of amplitude but also to take into account the psychological side of salience led to new approaches. The most widely used strategy for determining the salience of various peaks (at least in the spectral community) is an algorithm developed by German psychoacoustician Ernst Terhardt, which was implemented at IRCAM by Dan Timis and Gérard Assayag as *iana*. (It was later ported into real-time by Todor Todorof.) The details of the algorithm are beyond the scope of this article, but the goal is to refine the selection of peaks, taking into account both physical principles (critical band, frequency response of the ear, etc.) and cognitive ones (construction of virtual fundamentals); ranking the salience of each peak and eliminating peaks that may be loud, but that are masked by other partials. This technique has allowed composers to better deal with technical or orchestrational limitations on polyphony, selecting the best partials — even when they are few.

partial tracking

A more recent technique for data reduction, which is particularly well suited to large series of dynamic FFTs is partial tracking. This technique looks for connections between successive analyses, trying, in essence, to connect the dots. This generates musical lines from the series of analyses. It is generally used for resynthesis applications, but has also been used as a model for instrumental realizations.

Rhythmic Concepts

With the rhythmic and formal concepts described in this and the following section there are many fewer ideas described than in the previous sections. On a formal and rhythmical level spectral music is more continuous with other musical trends of the late twentieth century; whereas, it is more divergent on the level of pitch, harmony and timbre. I will limit myself to ideas which are both important to spectral music and particular to it. Ideas which are part of the lingua franca of contemporary composition will not receive specific treatment here.

absolute duration vs. symbolic rhythm

Just as frequencies offer spectral musicians a more direct access to many sonic structures than notes do, absolute temporal durations are often an easier way to conceptualize time and rhythm than the symbolic subdivisions of musical notation. This continuous conception has been less widely exploited for rhythm than the equivalent one has been for frequencies, since the problem of approximation is greater and the accuracy expected from performers and perceived by listeners is much less. Therefore the domain in which durational rhythmic thinking has been widely applied is limited to macro-rhythmic relations along with a few special case relations, in which durations have great advantages. In these situations durationally conceived relations are often more flexible than symbolic ones. An identical temporal structure can easily be stretched or compressed and can have the number of events increased or reduced without changing the framework of its overall perception, whereas this is often difficult or impossible in a traditionally notated passage without completely re-notating it or changing the tempos (which in certain contexts may not be possible or desirable).

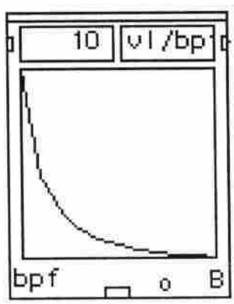
quantification

The approximation of continuous temporal events into the discrete units of musical rhythmic notation or any such units is usually referred to as quantification. This name is an explicit reference to the sort of grid that is presented by rhythmic subdivisions. Unlike the grids dividing frequencies into the nearest available pitch, rhythmic grids are hierarchical; this makes convincing quantifications difficult to produce. Also, the ear's ability to discern rhythmic anomalies is very context dependent. For example, during an accelerando, if one of the durations is longer than its predecessor (a micro rallentando) by even a few hundredths of a second, it will be perceptible; whereas a note that lasts eight seconds and one that lasts nine seconds will be indistinguishable when there are no external beat cues to mark the length. The result is that, unlike frequency approximation, there is not one closest and best approximation for each resolution (semi-tone or quarter-tone, etc. for frequencies, maximum beat subdivisions for rhythms), but different sets of compromises that must be made in consideration of the musical context and constraints. This often leads to the impossibility of using automatic quantifications that have not been manually modified by the composer. As a result, many of the passages in spectral pieces whose rhythmic conception began in the domain of durations were either quantified by hand or were 'touched up' manually. A system was developed recently at IRCAM by Carlos Agon, Gérard Assayag and myself which seeks to redress this situation by providing an interactive quantification environment called 'Kant,' where compositional concerns and calculational strategies can coexist in a user-responsive environment. This system allows the composer to work towards an acceptable quantification rather than simply offering a single, perhaps unusable, one.

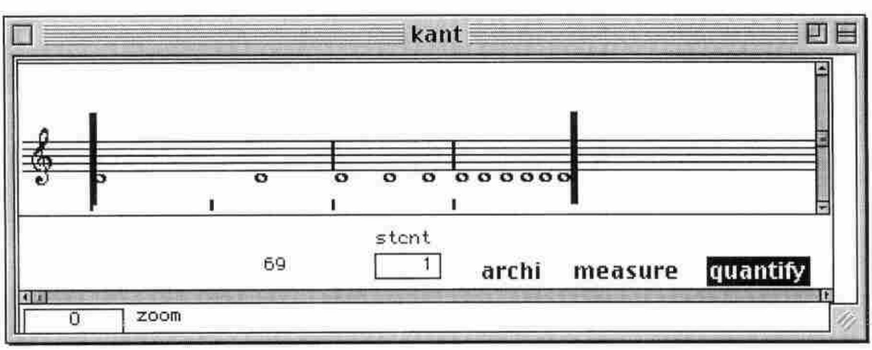
accelerations/decelerations

Among the structures most often modeled as durations in spectral music are accelerations and rallentandi. In order to produce a psychologically convincing impression of speeding up or slowing down, the duration changes must be exponential, not linear. This makes them particularly well suited to approximation by curves. These curves present intuitive graphical representations of the speed changes, that can then be adapted and quantified to fit diverse musical situations. While convincing accelerandi and rallentandi can certainly be written directly in rhythmic notation, they are extremely difficult to alter in this format (for example by adding events within the speed change); whereas curves are extremely malleable and allow the composer to generate entire classes of solutions which create the same sense of speed change.

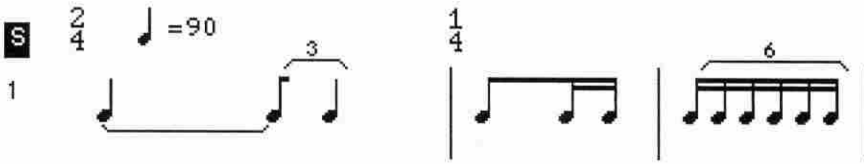
The following example is an acceleration curve, which is generated by an exponential function, then edited and displayed as the following curve:



The curve is sampled at discrete points which are then proportionally represented within the Kant quantification editor:



The results are finally quantified into standard rhythmic notation:



Especially noticeable in a very simple quantification, like this, is the loss of information. While we have no rhythms longer than those which precede them, the continuous nature of the acceleration has been lost. In a given context this might or might not be a problem. The only solution short of a tremendously complex rhythm, which is unlikely to receive an accurate rendering, is to adopt some element of proportional notation (like accelerando beams). It is precisely this type of problem which has led spectral composers to mix proportional and traditional rhythmic notation in their works. The attitude used in most spectral works is purely pragmatic: that which can be most clearly expressed through traditional notation uses it, while that which can only be awkwardly or inexactly expressed through this notation turns to the alternate means of proportional or otherwise personalized notation.

models from electro-acoustic sources

There are many other types of durational models that have been used by various composers. Other than curves, the most important class of models are the ones inspired by electro-acoustic procedures like echoes, delays, tape loops, etc. This type of manipulation is simply the application of widely used techniques from tape music into the realm of rhythmically notated instrumental music (via quantification).

models from sonic analyses

As was the case with harmonic structures, the initial use of relatively simple mathematical models, such as those described above, has been enriched in recent years with more complex models extracted from actual sounds through dynamic frequency analyses. Many different kinds of rhythmic information can be extracted from these analyses (whether it be dynamic contours, spoken rhythms or the pacing of timbral evolutions, etc.). Composers have extracted this information from all sorts of sounds — from crashing waves to recited texts to instrumental gestures. These models do not just offer rich harmonic materials, but also propose very interesting rhythmic structures, which can be used along with the corresponding harmonic material or independently from it. Of course, the translation of the often minute rhythmic fluctuations found in nature into an instrumental context requires a very prudent control of the quantification and a judicious mixing of traditional and proportional notation.

rhythmic distortions

The high degree of perceptual clarity, and the accompanying predictability, that many durational rhythmic structures offer has led many composers to distort these structures by varying degrees. These distortions echo those which are performed on harmonies. The simplest types of distortion preserve the relative lengths of the rhythms; a good example of this is rhythmic stretching or compression, whereby the relative lengths of each duration is preserved while the total duration is augmented or diminished (this is identical to one of the procedures for harmonic distortion mentioned above). Another technique is to add a percentage of random rhythmic fluctuation to the durations. When this percentage is small, the underlying structure is still very present, but the surface is made less predictable; when the percentage is increased, the randomness can begin to overpower the underlying structure. Combinatorial permutations are also frequently used to disturb the linearity of a rhythmic model (for example, swapping the positions of two events within a sequence of ten events can provide a moment of surprise and contrast within a strongly directional sequence). These distortions greatly increase the flexibility that can be achieved with a relatively limited number of rhythmic models.

Formal Concepts

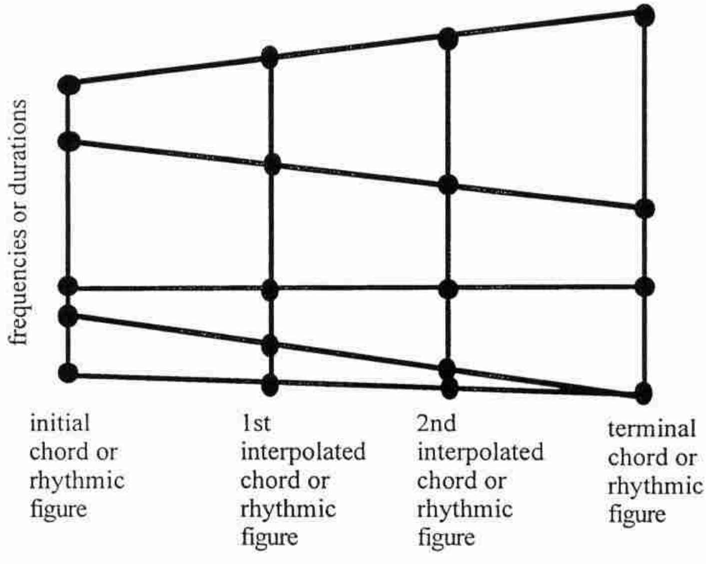
process

While not unique to spectral music, the idea of continuous transformation from one state to another, or process, has taken on a special manifestation and played a crucial role in the formal construction of spectral music. The types of processes found in spectral music are significantly different from those of minimalist music, for example, in that they affect all the musical parameters together, rather than acting on only one or two (like phasing). Typical processes of early pieces from the movement were transformations from order and stability (which includes harmonicity) towards the disorder and instability of noise, or vice versa. A good example of this kind of multi-parametric process is the beginning of Murail's piece **Gondwana**, where an orchestrally synthesized bell —

with the spectral profile, envelope and instrumental attacks necessary to achieve that structure — is gradually transformed into an orchestrally synthesized brass sound. The kinds of processes used in this music are distinct from many formal processes found in other types of music, in that they function on a perceptible levels. They are not underlying mathematical structures, but permeate all levels of the piece and are an important aspect of the perceived musical movement and evolution.

interpolation

The kind of smooth transformation from one state to another described above often makes use of interpolations. These are used in almost all aspects of the music, especially pitches and rhythms. The initial and final states are placed at the two end-point of lines or curves which are then sampled at various points to generate intermediate states. This type of procedure is especially effective in the continuous domains of frequency and time, but is sometimes performed directly on symbolic notes or rhythms. The following figure illustrates the construction of two steps interpolated along lines between the initial and terminal states (these lines could also be replaced by curves or other forms to produce a less directional result). It should be noted that the power of this technique is the novelty that can occur in the interpolated states. Interval or time structures that would have been impossible to abstractly imagine can sometimes grow out of these interpolations. In effect, the true interest is not the beginning or the end, but the discoveries that are made on the way.



limiting directionality — processes of processes

The main strength of processes and interpolations is the sense of direction and even inevitability that they impart to the musical evolution. This asset can, however, become a drawback, when it leads to predictability. One of the most significant changes that has occurred in spectral music in the last 15 years is the desire to find strategies which will reduce this predictability while still preserving the previously acquired directionality. One technique that has been used to this end is anamorphosis, which is based on the anamorphosis technique of medieval painters. The idea behind this technique is to present a single object from different perspectives, which distort the object in various ways — sometimes even making it appear to be a different object altogether. In this way, one object can develop into a rich reservoir of musical and formal material that can sound very different in spite of its high degree of relatedness; creating very different and surprising effects without compromising the coherence of the musical material. Another technique is to skip steps within a process. In a clearly directed process, the composer creates unpredictability, in the rate of change, and contrast, between the non-adjacent steps, through this procedure. As with anamorphosis, the overall direction of the process is strong enough to support these local contradictions without losing its power to provide global direction.

The most interesting attempt to subvert the single-minded directionality of the process is also the most structurally significant: building processes which use as their elements not harmonies or durations but other complete processes. For example, one might begin a piece with a process which has dense noise-like harmonies in low registers, rising and becoming more harmonic and more definite in their melodic and figurative content; this entire process might then be treated as a unit which is gradually bent until it becomes an opposite process of sounds sinking, becoming less harmonic and more diffuse gesturally. The complex intermediate states in this type of multi-layer process can often produce novel musical textures and situations. Once these processes of processes are combined with the other strategies of anamorphosis and incomplete presentation, they offer spectral composers powerful tools for building complex, unpredictable forms which, nonetheless, maintain cohesion and directionality.

Associated Topics

The topics in this section are not precisely about spectral techniques or ideas but are associated issues which have been important to many spectral composers. Due to the tangential relationships of some of these issues to our main subject, I will try to be very brief. I provide this information simply to orient readers unfamiliar with these ideas; those with a deeper interest should look to publications more focused on these topics.

Computer Assisted Composition

Spectral music uses many procedures which require calculations (such as the frequency calculations described in the first section of this article), but it is not truly algorithmic music. The calculations are required to generate basic material (even the most basic conversions from frequencies to notes, for example, can be very time consuming when performed manually); this material is not, however, used directly, but is manipulated musically by the composer. When these calculations represent a significant investment of time, it is difficult for composers to feel free with the material generated. They are unlikely to throw away weeks of elaborate calculation just because it is not exactly what they sought. They are more likely to perhaps tweak it a bit and then make do. Yet this freedom to experiment and to evaluate (even extremely complex) material is exactly what the spectral composers needed. The timing was fortunate, in that computers were beginning to become prevalent and their usefulness for this application was evident. For the computer, none of these calculations were of significant complexity and, thus, with the proper environment a composer could work freely and intuitively with a material of almost any complexity. This was also useful for demystifying calculation. Once the hours, days and weeks were converted into seconds, it became harder to attribute any abstract significance to the mere fact of calculation and returned the emphasis to the real end product: the musical result. The earliest Computer Assisted Composition (CAC) programs were of relatively limited scope. They were created for or by single composers or small groups. Two notable examples were programs written by Tristan Murail (in the early 1980's) and another program that regrouped various musical functions used by a few composers at IRCAM in the package Esquisse (in the late 1980's). The first effort to create a more general environment which would allow many composers to develop the personal environments they would need for their work was the program PatchWork (first conceived by Magnus Larson, then developed at IRCAM by Camillo Rueda, Gérard Assayag and Carlos Agon). In the earlier and mid 1990's, this program spread the use of CAC far beyond previous levels and has also allowed spectral composers to find much more freedom in their daily compositional work. This program is essentially a graphical musical programming language, which gives the composer the power to make personal functions and environments. OpenMusic, a successor to PatchWork with increased capability and better graphics, is now in development at IRCAM.

AudioSculpt

The ability to perform spectral analyses used to be the province of universities and research centers. Composers would keep collections of results which they could consult. The Macintosh program AudioSculpt, developed at IRCAM by Chris Rogers and Peter Hanappe in the mid 1990's, has changed this situation. AudioSculpt gives access to sonograms, spectral analyses with or without Terhardt's salience algorithm, simple partial tracking, and easy connections for using the data produced by AudioSculpt within the computer assisted composition environment PatchWork. This allows composers to generate and manipulate complex spectral materials at home, in a format that integrates well with the rest of their compositional environment.

synthesis techniques

Spectral composers often use, either directly or by analogy, different techniques developed for sound synthesis. Therefore, it seems necessary to give brief definitions of a few of the most influential of these techniques.

additive synthesis

This technique creates complex sounds by combining simple sounds with different amplitudes. The mixture stops on organs are the oldest form of additive synthesis. While theoretically capable of producing any sound and conceptually attractive and intuitive, additive synthesis requires an enormous number of partials and complex control to produce even moderately satisfactory results. Thus this technique, in the past, was often used for the generation of relatively simple sounds. Now there are programs which generate the parameters for controlling additive synthesis based upon the results of analyses. These analyses can, of course, be modified before the sounds are re-synthesized. This powerful technique, called analysis/resynthesis, has caused a resurgence in the use of additive synthesis. As described in the first section, the concept of additive synthesis has been extremely influential upon spectral musicians.

subtractive synthesis

The opposite of additive synthesis, subtractive synthesis, begins with extremely complex sounds, often white noise (sound with equal energy in all frequencies), which is then filtered to leave only the desired portion of the sound. This technique, also theoretically capable of producing any sound, is easier to use than additive synthesis: since non-controlled parameters already exhibit complexity and, thus, all sonic richness does not need to be precisely specified. Conceptually, however, this technique is less intuitive and the results, while richer than those produced by additive synthesis, are often much harder to control. Spectral composers have made relatively little use of this technique.

FM synthesis

The technique of FM synthesis, developed by John Chowning, has been widely used because of its ability to produce fairly complex sounds with relatively little computational cost and relatively simple control parameters. The spectral content of FM sounds was explained in the first section of this paper. What must be noted here is that the FM synthesis technique is not (even theoretically) capable of producing all possible sounds. They create a large but finite class of sounds, which often have a certain family resemblance. This class of sounds is very present in the ears of many spectral composers, through Yamaha's DX series synthesizers which were based on it. It is especially successful for metallic sounds but much less so for strings.

sampling

The most common technique in newer commercial synthesizers is sampling. This technique uses tiny bits of recorded sound (samples) which are the replayed with modified envelopes, filtered, stretched, looped, etc. to create the final sound. This technique has provided little in the way of conceptual ideas to spectral composers, but as a common technique it has often been used in the electronic portions of their pieces.

tape loop, re-injection loop, echo and delay

These techniques are all based on the recording of a sound followed by a playback of that sound after an interval of time. The oldest technique was the tape loop. A loop of tape was passed between two open-reel tape machines, one for recording and the other for play-back. The distance between the machines and the speed of the tape determined the amount of delay. When the input is an open microphone, this is called a re-injection loop, since the play-back is re-injected into the recording through the mixing board: producing a proliferation of sound. These techniques are now generally modeled digitally — with the delay replacing the tape loop and the echo replacing the re-injection loop. The advantage to these digital units is a finer control (for example, the precise amount of sound re-injected into the echo can be regulated with a feed-back control) and much better sound quality. The concept underlying these procedures — a sound copying itself to generate a sonic structure different from the original sound — has influenced many spectral musicians.

software source

Many of the computer programs described above were developed at IRCAM and are available through their Forum users group. Those desiring more information can contact IRCAM through their web page at http://www.ircam.fr or by mail (Forum IRCAM, 1 place Igor Stravinsky, F-75004 Paris, France).