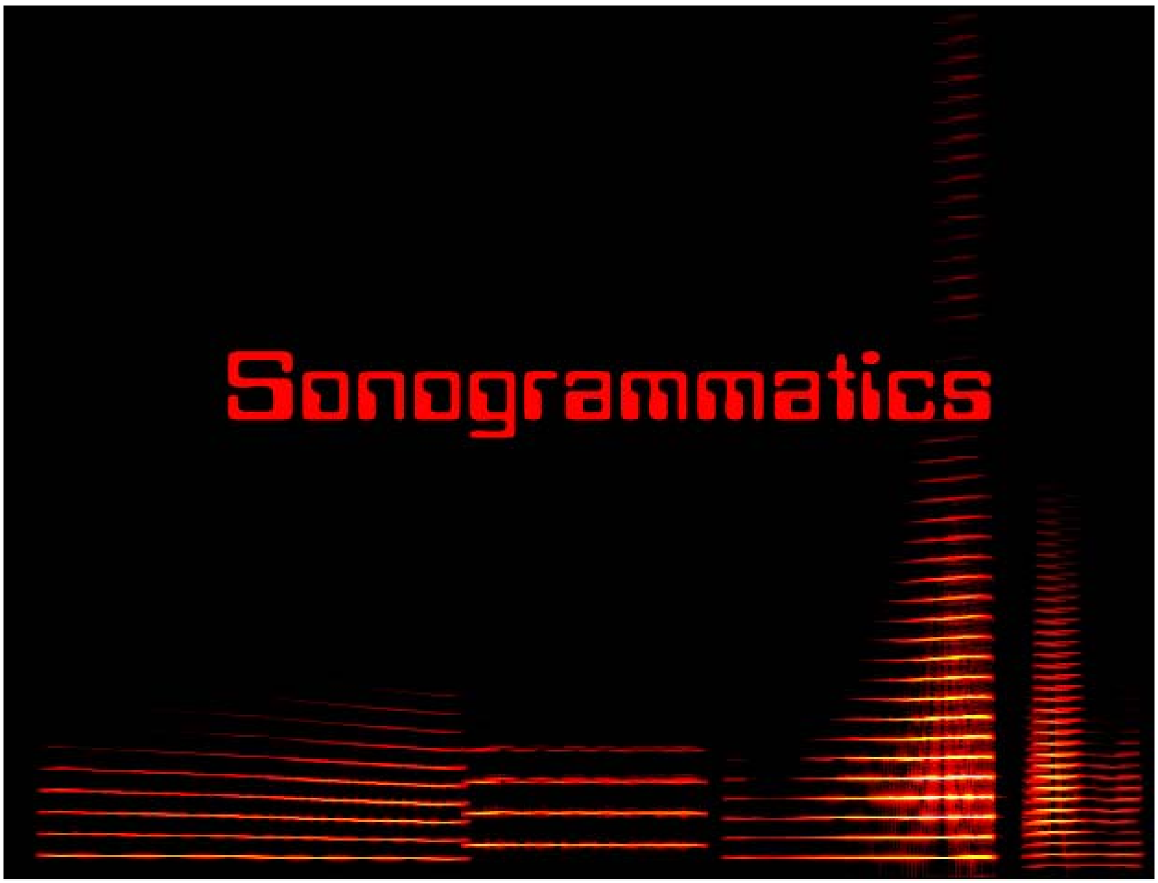

A Method of Musical Composition Based on Sound Analysis

by

Jeremy Starpoli

Division III Hampshire College

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I. Introduction

Making music can be defined as the means that people use to organize sounds into unified compositions and improvisations, thus one of the goals of a musician, whether performer, composer or listener, is to learn about the nature of sounds and how they are used in musical works. Each individual organizes sound in their own way, from conscious listening to rigorous virtuosic creation of preconceived sounds. The methods of learning about sound are also varied, each valid in its own context, and each can be mined for its usefulness in developing new forms of organizing sound on a personal level. To be a musician (again in the broad sense), especially in this age of extensive global communication and historical records, offers the opportunity to discover and study the musical systems of people throughout the world and across time. When sifting through the wealth of written material about music, and when listening to music itself, how does one condense this information into a personal approach that is consistent with one's own musical ideas and goals? In this essay, we will attempt to form a method for doing this, and will draw on the work done by a wide range of performers, composers, theorists and scientists in a variety of styles and fields. Sonogrammatics is the invented name for this method, meant to imply the "grammar of sound" as well as the scientific precision found in the field of musical acoustics, where the visual representation of sound as a sonogram is commonly used. The method provides a set of tools for musicians, culled from the analysis of existing music, and these same tools can be isolated and put to use in developing diverse musical works through composition and improvisation.

The common denominator of all music is sound, therefore any attempt to develop a holistic approach to the analysis and creation of music necessarily will have to examine what sound is at a fundamental level. While there have been many descriptions of sound, ranging from poetic to mystical to allegorical, the scientific field of acoustics provides insight into its physical nature. By studying how a musical instrument makes the sounds it does, the field of musical acoustics corroborates what is often described less precisely in musical work itself. Most musicians and acousticians would agree that there are many parameters that make up even a single sound, and acoustics gives us a precise terminology for describing those parameters. Specifically, a sound has *pitch* (frequency), *loudness* (amplitude), and *timbre* (spectrum and envelope) and any or all of these can be varied over *time* (duration). In Part One of this essay, we will examine the acoustical definitions of these four parameters, and how they are represented in musical practice, and thus we can develop a holistic method of describing almost any single sound. A single sound is one that comes from only one source, like a single key of the piano, as opposed to a chord of multiple strings sounding at once. A full-scale method for the analysis of multiple simultaneous sounds is beyond the scope of this essay, yet most multiple-voice instruments are also capable of producing single sounds. Therefore, the methods proposed here can serve as a basis for further research into the complex issues involved in analysis of simultaneous sounds. It is intended that Part One serve as an introduction to the most useful (in my own experience) scientific concepts and measuring scales that have been developed to describe the parameters of pitch, loudness, timbre and time.

Another feature that is basic to our perception of single sounds, and useful in musical analysis and composition, is the distinction between constant, gradient and variable sounds. A *constant* sound is one where all of the sonic parameters remain static throughout the sound's duration. This includes sounds that waver between points in a continually repeating fashion, like a trill or vibrato. A *gradient* sound not only changes from one point to another (in one or more parameters) over the course of its duration, but also does so in a continuous manner, smoothly connecting the two points. Of course, it is also to create sounds that change in some parameters over time in an irregular fashion, and these would be classified as *variable* sounds. One of the purposes of this essay is to propose that all of the sonic parameters can have equal importance within a musical work, thus constant, gradient and variable sound types become the fundamental "building blocks" of such an approach. This is not to say that every piece composed by these means must contain equal numbers of these sound types, but that the functions of all of the parameters of sound are taken into careful consideration when designing a musical work.

Once we have an idea of what sounds are comprised of, and how sounds can be classified according to each of the four parameters, we are able to examine how these sounds are combined to make musical compositions and improvisations. Part Two of this essay attempts to define a set of *operations* that can be used to vary sound in each parameter. In our contemporary musical culture, these operations have been best defined in terms of pitch and rhythm (duration structure), and some of those principles can be applied to loudness and timbre as well. When a series of sounds is varied by one or more parameters, a *sequence* of sound events is created. Moreover, there is also a set of operations that may be defined for varying sequences into phrases, phrases into sections, and sections into whole musical pieces (i.e., the creation of compositional forms). Since there are so many ways of combining sounds, the set of operations that can be used to do this is continually expanding. For this reason, Part Two will only touch upon the most essential. However, the definition of these micro- and macro-level operations provides us with a method for both creating and analyzing a wide variety of musical pieces. The analysis of changes in the sonic parameters over the course of a musical piece can give a sense of the operations used to vary the single sound events, regardless of the style or genre of the piece itself. This can lead to a method of composing and improvising music by determining the operations that will be applied to the sounds of musical instruments.

As a science, acoustics has gained its vast knowledge by empirical means, experimentally testing the properties of sounds and our perception of them. While this practice, over the course of many centuries, has generated much valuable insight into the nature of sound, work in acoustics, like any science, is far from complete. For this reason, a major portion of this essay will be based on actual empirical research into the acoustics of musical instruments to determine their capacity to negotiate the four parameters of sound described above, a task not yet attempted in depth. A musical instrument is only a tool that performers use (and composers call for) to create sounds, yet each instrument, and its player, has unique capabilities that must be understood (on some level) to be utilized properly. Many of the concepts in this essay were developed by a studying group of eleven acoustic instruments empirically: trumpet, trombone, tuba, flute, saxophone, oboe, viola, guitar, piano, vibraphone, and drum set. From this set, at least one of each of the major families of acoustic instruments is represented, and sometimes more when special characteristics were worth studying (brass and woodwind aerophones, for example, or bowed and plucked strings). The sounds samples recorded from each instrument were analyzed on a computer to determine how the acoustic definitions of sound parameters relate to musicians' actual negotiation of them. The resulting information is integrated throughout the text and in the form of visual examples explaining the parameters of sound.

Part Three of this work is devoted to giving practical examples of how the method set up in the previous sections can be applied in actual compositions and performances. For the contemporary instrumentalist, the scales of measurement found in acoustic definitions of sound suggest new methods of training the ear to hear important attributes of sonic parameters. This can be applied to developing techniques to perform changes in these parameters with much greater precision than is common. Examples of beneficial tools and methods for developing these techniques are found in Part Three. An understanding of musical acoustics also provides insight into the types of sound available on a given instrument and how these might be produced. Also, an increased awareness of the operations that may be applied to the parameters of sound can be informative in the analysis of musical works for performance. In many pieces, parameters left unspecified in a score provide the means for the performer to add nuance to the notated material, and knowledge of parametric operations can suggest creative ways of doing just that.

To the composer, the ideas expressed herein provide tools that summarize and add to current musical practice, and a means of investigating the vast array of sounds available with certain musical instruments. By applying the transformational operations described in Part Two to specific formal structures, solo compositions for trombone were created. The process of writing these pieces is also described in Part Three. The motivating concept behind these compositions was the development of the distinction between constant and gradient sounds in a variety of contexts, including repetitive and non-repetitive structures, thematic development, and improvised parameters. Some of the music discussed in Part Three was written using standard musical notation, but a new system of notation was also developed that was designed expressly for a "sonogrammatic" music of variation in all sonic parameters. While each of the solo pieces focuses on a single compositional idea, ensemble music was also written using a variety of compositional techniques within distinct sections. The trio piece that is analyzed in Part Three outlines some of the issues involved in combining approaches within a musical work, and points in the direction of future research into the acoustic analysis of simultaneous sounds

The scope of this essay is relevant and applicable to many types of musicians or even those simply interested in how sound works and what makes a musical piece. Although the voice and electronic instruments have been left out of the present study (due to the sheer number of possible combinations of timbre and/or loudness), the information that is provided in this essay is generally applicable to any type of sounds and how they are arranged. In this respect, parts of the essay may be of interest to untrained readers interested in the properties of sound and how they are arranged into compositions. Of course, familiarity with terminology from the fields of acoustics and music could be useful to the reader, but they are not required, since most of the important terms have been defined in the text.

By abstracting the forms and materials of music, Sonogrammatics, like other systems of music analysis, gives an substantial means of examining a sound's physical and perceptual attributes (how it sounds), and what its function is within a local and global context (why it was chosen). Traditional ideas of consonance/ dissonance, or "good sounds"/ "bad sounds", which are likely outdated anyway given the wealth of recent criticism of these models, are less important to the present writing than the methods by which a piece of music is created, to which virtually any sound might apply. That said, when moving from the relatively passive mode of analyzing an existing piece of music to the more active mode of creating one, this method is rooted in the "radical" sonic ideals of the 20th-century Western European tradition. The concept of music based on parameters other than pitch or rhythm stems from this movement, and many of the issues that this concept raises have been developed more extensively since then. The music that one makes using the Sonogrammatic method may sound radical to those unfamiliar with other experimental compositional theories, yet the method is sufficiently varied that many types of musical forms and sounds can be developed using its techniques.

Sonogrammatics attempts to extract from various sources the means of thinking about the general processes involved in music and analyzing the specific details of how these are carried out. Each source cited in the text provides certain insights into the nature of music but also concepts that are not always relevant. For the purposes of this essay only the beneficial concepts from a source will be discussed, unless it is useful to point out any deficiency that detracts from the implementation or understanding of a concept. Any system of analysis must necessarily reduce the amount of information studied in order to emphasize the importance of certain aspects, and in this way Sonogrammatics draws on the general field of music analysis, while incorporating the more precise terminology of acoustics. While there is a thorough examination of the major parameters of single sounds within this essay, it is not meant to be a complete catalog of all aspects of sound and its transformation. It is hoped that, through the scrutiny of the concepts herein and their development into actual musical work, Sonogrammatics will prove to be adaptable to new concepts while retaining its disciplined approach to investigating music from the smallest sound event to the complete musical work.

Part One: Analyzing the Parameters of Sound

1. Pitch

The first parameter of sound that we will examine is that of pitch, and this decision is not arbitrary. In many musical cultures across the world and throughout time, pitch is *the* fundamental parameter of musical sound, and, in some cases, all of the other parameters are considered secondary. In our own contemporary musical culture, the predominant means of notation and (music theory) analysis are based on the parameter of pitch, with rhythm playing a close second, and dynamics, timbre, etc., primarily conceived as effects applied to a pitch. In attempting to examine each sonic parameter individually, it makes sense to start with the parameter that has had the most practice being abstracted (isolated) from the others. Moreover, having a working knowledge of pitch will be most useful in examining the other parameters of sound, especially timbre, as we shall see. Once we have defined pitch, we can then analyze the other parameters in relation to pitch.

The word pitch is used here to speak of the quality of a sound that determines how "high" or "low" that sound is. The field of acoustics gives us a more precise definition of this quality, which is a property of the vibration of particles in a medium. A sound created in air is a vibration of the air particles themselves, acted on by the expenditure of energy. This energy displaces the air particles immediately around it, causing them to crowd the adjacent particles, thereby increasing the pressure. When the original particles return to their position, this causes the second set of particles to follow, which decreases the pressure in that area. This alternation of compression and rarefaction causes a wave, much like the ripples of a stone thrown into a pond. The sound wave expands out into air in all directions, and loses energy by being absorbed by surfaces, eventually dying out unless more energy is expended at the source. If we graph the displacement of energy over time, we can view the *waveform* of a sound, which is one of many useful visual representations in sound analysis (**Fig. 1.1**).

For a sound to be heard as a pitch, it must consist of many cycles of a sound wave, and one cycle of the wave is the return of a particle to its origin after

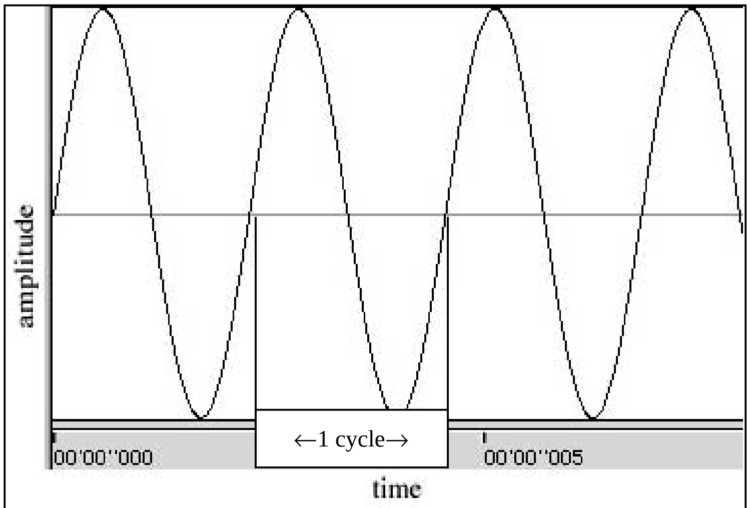

Figure 1.1- waveform of a sine tone.

being displaced in both positive and negative directions. The number of these cycles per second determines the *frequency* of the wave. The standard unit of measurement of frequency is the *Hertz* (Hz), a term synonymous with cycles per second. If a cycle repeats itself exactly over a specific time interval, known as the *period*, then the resulting wave is called a periodic wave. Mathematically, the simplest example of a periodic wave is a sine wave, which theoretically repeats indefinitely over time, therefore having a constant frequency. A sine wave that has a frequency of 440 Hz, for example, causes the air particles around it to be displaced from their original positions (in both directions) 440 times in one second (Campbell and Greated, 1987).

Up until the advent of electronic technology, the sine wave (or "pure tone") could only be approximated, but since then much research has been done involving our perception of sine waves. It has been found that the human ear perceives vibrations in air as sound between the frequencies of (roughly) 20-20,000 Hz. This is the *pitch spectrum* (not to be confused with the spectrum of a single sound). Research has also found that a tone that has a lower frequency number is perceived as being "lower" in pitch than one having a higher frequency. It can also be said that this lower pitch has a longer *wavelength*, or distance between the crests of a sound wave in air, since the fewer cycles a wave completes per second, the longer it will take one cycle to be complete. However, it should be noted that this only applies to pure tones. Combinations of sine waves that are close together in frequency can have long waveforms even though they are perceived as a single sound of relatively high pitch, yet these sounds are not "lower" than they are heard to be (Sethares, 1999). A *noise* is defined as a sound whose waveform does not repeat, and consequently has a variable wavelength. Depending on the range of the spectrum covered, noises may also give a sense of pitch. Our perception of pitch, then, is often directly related to the actual frequency of the sound vibrations in air, but it should not be assumed that this is the only criterion. Pitch is a subjective sense based on many factors, including but not limited to: frequencies, wavelength, periodicity, the presence of vibrato, age and training of the listener, etc (Campbell and Greated, 1987).

As further evidence that our sense of pitch is a subjective one, consider that most sounds, other than some generated by electronic means, consist of more than one frequency and are called *complex tones*. Most acoustic musical instruments (except percussion), for example, in their normal playing mode generate pitches whose component frequencies (approximately) form what is known as the *harmonic (or overtone) series*. The structure of this series is such that each successive frequency, called an overtone or partial, is a positive integral multiple of the *fundamental* frequency. For example, if the fundamental frequency of a (harmonic) complex tone is 100 Hz, then it also has overtones occurring at 200 Hz, 300 Hz, 400 Hz, etc. If we look at the relative amplitude, or loudness, of the frequencies present in some instant of sound, or the average frequencies over a sound's duration, we are then looking at its *spectrum* (see **Fig. 1.2**). A sound's spectrum plays a major part in determining its timbre or sound color, as we will see in Section 3. For the purposes of discussing pitch, though, we need only note that the presence of a set of frequencies where the components are integral multiples of a fundamental frequency (which need not be present) indicates that a single pitch can be perceived, roughly corresponding to the fundamental (Plomp, 1967).

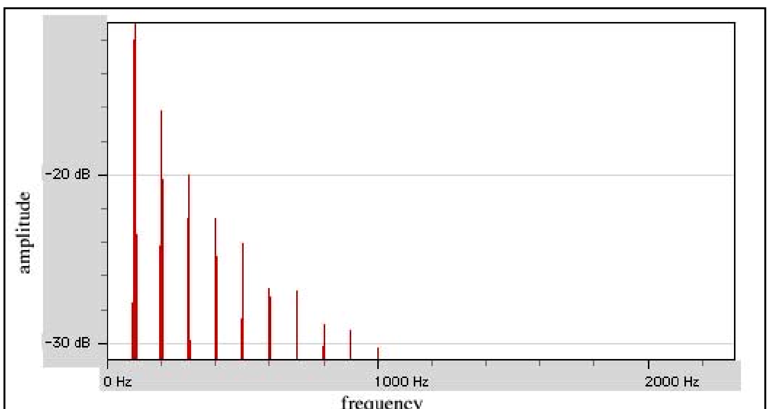

Figure 1.2- the spectrum of a harmonic tone.

Non-harmonic tones

Just as pure sine waves rarely occur in actual musical practice (except when generated electronically), a perfect harmonic series is likewise difficult to find. Non-harmonic sets of frequencies can also fuse into a single perceived pitch as well. The harmonic series can be stretched or compressed a small distance (about 1/12th of an octave, in either direction) and the resulting pitch will still function as a *quasi-harmonic* tone (Sethares, 1999). Also, many percussion instruments, such as drums or bells, generate overtones that are non-harmonic, and yet a single perceived tone is heard. Although researchers are still uncertain as to exactly how the ear and brain respond to non-harmonic partials, the current theory is that the ear searches for overtones within a dominance region (roughly 500-2000 Hz) that form an approximate harmonic series. The brain then responds by hearing a pitch with that fundamental. There is a certain ambiguity to the pitch, depending on the number of partials, for the more partials present, the more approximate pitches that might be perceived (Campbell and Greated, 1987).

Noise

If a sound source emits a wave that fills a certain part of the pitch spectrum with frequencies, then that sound is called a *noise*. However noises can also be "pitched", depending on how wide the band of frequencies is. White noise covers the entire audio spectrum with equal amplitude, and is therefore not pitched, but pink noise covers only a limited range (Roads, 1996). Some percussion instruments, such as cymbals and bells, and some techniques of playing normally harmonic instruments (like flutter-tongued winds) produce noise sounds (Erickson, 1975). Noise is easy to produce and filter to specific regions of pitch with electronic instruments as well (Roads, 1996).

It is often difficult to discern where a non-harmonic, partial-rich, complex tone becomes a noise, or where a stretched harmonic tone becomes inharmonic. Perhaps there is another spectrum of pitch besides the one from 20-20,000 Hz.

And that would be the spectrum of how a sound is perceived as a pitch, ranging from pure sine waves, to fused harmonic tones, to non-harmonic tones, to noises (**Fig. 1.3**). Where a sound falls within this spectrum indicates the relative "pitchedness" of the sound. Adaptation of Sethares' "dissonance meter", or other measurements of a sound's harmonicity, could prove helpful in determining where on the perceived pitch spectrum a sound lies (1999).

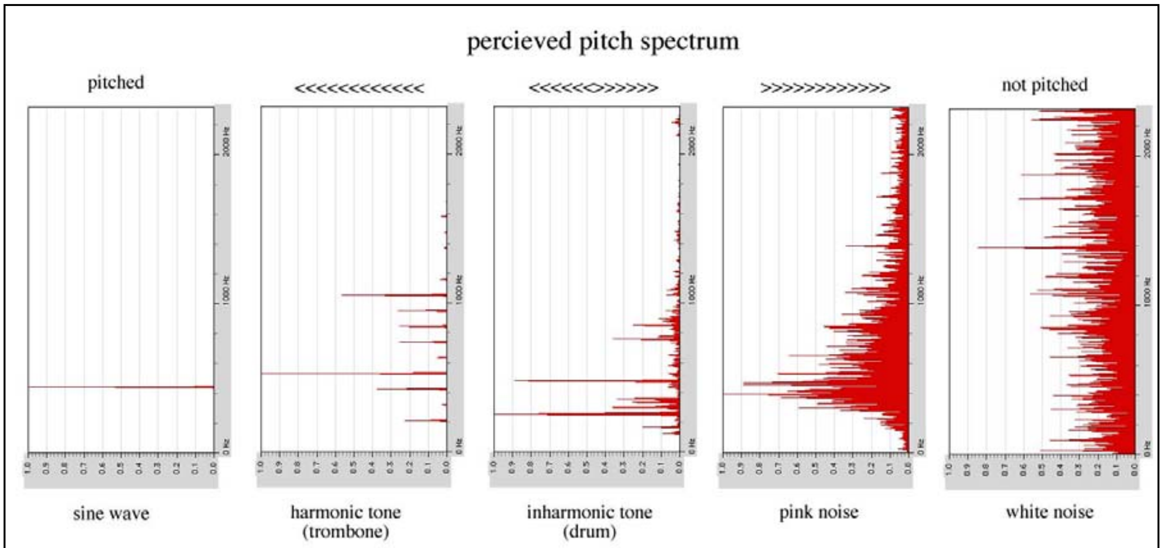

Figure 1.3- perceived pitch spectrum, with highly pitched spectra on the left side, progressing to unpitched noise on the right.

<u>The Octave</u>

Now that we have a basic understanding of what pitch is, and how it works, let us delve even deeper into how pitches are combined to make music. Most musical instruments are capable of sounding more than one pitch, either consecutively or simultaneously, so what happens when they do? In essence, a relationship is set up, using one tone as a reference point, and measuring how far away the second tone is. This distance is called an *interval*, and one way of naming intervals is by showing the relationship between the two tones as a fraction, or *frequency ratio*. Placing the higher frequency above the lower and reducing the fraction to lowest terms creates a frequency ratio, which is a useful tool for expressing how one pitch relates to another (Campbell and Greated, 1987).

The earliest investigations of the nature of pitch intervals were done by ancient Chinese and Greek men of learning, who studied pitches by dividing lengths of bamboo pipes and stretched strings, respectively. Both cultures found, independently of each other, that if you divide a string or pipe in half, the sound from the shorter pipe or string will be the *octave* (Partch, 1949). In the modern era scientists were able to determine that this procedure will double the frequency of the first tone, or, if one doubles the frequency, the octave is produced. In simple terms, then, an octave can be expressed easily by the ratio 2/1, and each successive octave must be obtained by multiplying the frequency number of the higher tone by two. The octave is significant to music based on harmonic tones because, in most musical traditions, a pitch that is an octave higher or lower than another tone is effectively heard as the same note in a different register. This is because many of the overtones of a pitch that is an octave of another tone line up with the overtones of that first tone, which increases the similarity between the two pitches (**Table 1.1**). In fact, since the first overtone in a harmonic series is twice the frequency of the fundamental, these two frequencies are separated by an octave, and all of the overtones that are multiples of two are octaves of the fundamental. A harmonic tone that is an octave of another tone is said to have the same *chroma*, which can also be called pitch color, or the property that distinguishes one pitch, in any octave, from another (Campbell and Greated, 1987; Burge, 1992).

Table 1.1 - frequency ratios of the overtone series of two harmonic tones, one an octave higher than the other, showing overlapping overtones.

frequency ratios of harmonic tone	1/1	2/1	3/2	4/2	5/4	6/4	7/4	8/4	9/8	10/8	etc.
frequency ratios of harmonic tone one octave higher		1/1		2/1		3/2		4/2		5/4	etc.

Register

The octave is not only useful for the analysis of tuning systems and intervals. It also helps us to define the pitch spectrum by dividing it into approximately 10 different *registers* (Cogan, 1984). Beginning with 16 Hz, about the lowest tone audible as a pitch, each successive octave defines a register:

Table 1.2- division of the frequency spectrum into 10 registers.

Register	1	2	3	4	5	6	7	8	9	10
Frequency (Hz)	16-33	33-65	65-131	131-262	262-523	523-1047	1047-2093	2093-4186	4186-8372	8372-16744

While frequency numbers give a clear sense of where in the pitch spectrum they reside, most tuning systems give names to each specific pitch within an octave. The names could be indications of the frequency ratios of that system (as in Just Intonation), or arbitrary letter names given to each pitch (as in Equal Temperament). In either case being able to state specifically which register a specific pitch is in is essential when analyzing musical passages.

Tuning systems

While the octave divides the pitch spectrum effectively, and defines a specific relationship between pitches, most musical systems throughout the world also divide the octave into smaller intervals. Such divisions are called *tuning systems*, and they are useful to musicians in many ways. First, for those instruments having *fixed pitches*, e.g., that have finger holes in a long tube (flutes) or many stretched strings struck by hammers (keyboards), for example, then a means of tuning each tone to a specific pitch is a necessity. In many musical cultures, there is a standard tuning system, or *scale,* that defines the intervals within each octave. This not only enables ensembles of fixed pitch instruments to all tune to the same scale, but creates a nomenclature for the pitches, which can be played by *variable pitch* instruments (like the violin or trombone) as well.

<u>Ratios and cents</u>

Before looking in depth at some specific tuning systems, we should first discuss the nomenclature of tuning systems, and how they are compared. First, let us point out that expressing intervals in ratios agrees with our perception of them. That is, intervals are proportional over the entire pitch spectrum, so that, perceptually, the distance between two tones of the same sounding interval increases as the frequencies become higher in the pitch spectrum. In current musical parlance, based on Equal Temperament, one interval (say, a major third) is added to another (a minor third) to create an interval that is the sum of those two (a perfect fifth). With frequency ratios, the ratios of the tones must be multiplied (5/4 X 6/5= 3/2) to achieve the correct overall interval (Campbell and Greated, 1987). Also, all major tuning systems use pitches that repeat at the octave, so that the intervals found in these systems all can be analyzed within one octave. Since 2/1 expresses the relationship of one tone to another that is effectively one half of that tone (either in frequency or string length, etc.), then any fraction where the smaller number is less than half of the larger number represents an interval greater than a 2/1. Such a ratio can be brought within an octave by doubling the lesser number until it is more than half the greater (Partch, 1949). That is not to say that intervals greater than an octave (known as *compound intervals*) are "equal to" their smaller counterpart within an octave, any more than a pitch that is an octave of another pitch is the "same" pitch as the first.

One other key concept will aid in our investigation of pitch, and the comparison of tuning systems, and that is the system of dividing the octave into 1200 *cents*. Cents are the unit of measurement that divides the octave into 1200 equally spaced parts, where the ratios of any two adjacent units are the same. This system was devised by Alexander Ellis to have a convenient means of comparing the tuning system Equal Temperament, which divides the octave into twelve equal semitones of 100 cents each, to other tuning systems (Helmholtz, 1885). The system is useful in more than one way, however, and one of its benefits is that it provides an easy method for analyzing the relative size of any intervals whose corresponding frequency ratios involve large numbers. Yasser, in his Theory of Evolving Tonality (1932), gives the example of the comparison of a "true" perfect fifth, with a frequency of 3/2, and its equally tempered counterpart, whose ratio would be 433/289. At first glance it is difficult to discern the comparative size of these two intervals, and even applying a common denominator (867/578 and 866/578) does not give a clear sense of the difference between the two. Note that actual frequency numbers representing these intervals would not remain the same if the interval were started on different frequencies. Yet if the two intervals are expressed in terms of how large they are in cents (3/2=702 cents, 433/289=700 cents), then one can not only see how large the intervals are comparatively, but also the difference between them (2 cents).

Both the cents scale and frequency ratios divides the frequency spectrum logarithmically. In this way, the interval of 3/2 is 702 cents whether it is between 56/84 Hz, or 1000/1500 Hz. To find the interval size, in cents, between two pitches one can determine the logarithm of the difference between the two fundamental frequencies, thus: {PI (pitch interval) = 3986 $\log^{10}(\frac{f2}{f1})$} . This system, while useful for comparing the intervals of tuning systems, however, is limited by not showing the difference between the two tones of the interval. If one knows the size of an interval in cents and one of the two frequencies, then it is easy to determine the ratio, and hence the second frequency. For this reason, the methods of analyzing intervals by cents and frequency ratios exist side by side in any study of tuning systems.

One other point about ratios and cents and their application to analyzing tuning systems bears mention. In analyzing tuning systems where the pitches are related to each other in relatively small number ratios, then those same ratios can be applied to naming the pitches themselves. In common practice, each tone of the scale is given a ratio relating to the starting pitch, within the 2/1. However, as Blackwood (1985), Partch (1949) and Yasser (1932) all point out in their works on tuning, this system does not transfer well to Equal Temperament or any other tuning that is based on equal divisions of intervals, as their corresponding ratios become unwieldy. For this reason, and for convenience of those familiar with traditional music theory, I will use the system of letter names in current use, but *only when referring to the pitches involved in equally tempered tunings.* This system, which gives the letters A through G as the notes of the white keys on the piano and certain sharps and flats as the black keys, is useful in that it gives an easily recognizable name to pitches that would be difficult to work with as ratios. Using this system to name pitches in all tunings, as many scholars have done, creates confusion as to which pitches are actually able to be called by a certain letter, and where the range of that certain letter has its boundaries (Partch, 1949). In any case, it is important to be as exact as possible when referring to complicated pitch structures, and "it is not precise to use musical terms until exact definitions have been framed" (Blackwood, 1985).

Pythagorean Tunings

Let us now look at musical intervals and how they are set up into a tuning system. The ancient Chinese and Greeks did not stop their investigations of musical intervals with 2/1, but sought to divide their pipes and strings into many combinations to produce musical scales that were pleasing to the ear, or based on mathematical principles. According to Partch, the next logical step after exploring the capabilities of the relationship of two to one was to introduce the number three. The 3-limit is utilized by dividing a string or pipe into three equal sections, and one can then obtain the intervals 3/2 (the "perfect fifth"), and its inversion 4/3 (the "perfect fourth"). The use of this 3-limit as a basis for a scale (tuning system) is what characterizes the scale that the ancient Greek scholar and his followers propounded:

Table 1.3- Greek Pythagorean scale constructed with ascending 3/2s.

1/1	9/8	81/64	4/3	3/2	27/16	243/128

This scale is formed by using the pitch of six ascending 3/2s, and then bringing those pitches down within an octave (3/2 X 3/2 = 9/4 = 9/8, etc…). Consequently, all of the pitches are related to one or two others by a 3/2 or 4/3.

The idea of the bringing successive 3/2s within an octave to create a scale can be used in a variety of ways, and so Partch posits that any system which utilizes this may be dubbed *Pythagorean*, after the Greek philosopher who expounded this method (1949). For example, the ancient Chinese actually used this method in the formation of their pentatonic scale, which consists of the first four 3/2s of the above scale:

Table 1.4 - Chinese pentatonic scale.

1/1	9/8	81/64	3/2	27/16

(Partch, 1949). In order to transpose their melodies into different keys, the Chinese used each of the 3/2s as a starting point for a different pentatonic scale with the same qualities as the first. Thereby a twelve-tone musical system was created where only five pitches were used at any given time (Yasser, 1932).

There is still debate about the musicality of Pythagorean tuning. As Partch notes, the scale may be easy to tune, but that does not necessarily make it easy on the ears. He even questions whether or not the ratios of 81/64 or 27/16 can be sung by the voice without instrumental accompaniment (1949). In contrast to this viewpoint, there has been recent research that voices singing a cappella may actually gravitate towards naturally singing in Pythagorean tuning (Backus, 1977). In judging any tuning system, however, it is essential to understand the goals of the musicians using a particular system, as well as what effect it creates. This is the position of Sethares, who postulates that the intervals of a scale sound the most consonant when played on instruments having a timbre related to that scale (1999).

Just Intonation

The term *Just Intonation* (abbreviated JI), like Pythagorean tunings, refers to a general method of obtaining pitches, rather than a specific scale per se. The method used relies heavily on the ear as the judge of acoustically pure intervals, that is to say intervals whose ratios involve the smallest numbers (Partch, 1949; Campbell and Greated, 1987). In this respect, a JI scale is directly related to the overtone series, where the relationship of each overtone to the preceding one gives the small number ratios in order of decreasing size (i.e., 2/1, 3/2, 4/3, 5/4, 6/5, 7/6, etc). Most just-intoned scales utilize 5/3 as well, since the interval occurs in the overtone series. In essence, JI is any system that extends the idea of the 3-limit to the other odd numbers. In his Genesis of a Music (1949), Partch describes the evolution of the acceptance of higher number intervals as consonant, up to the 7- or 9-limit accepted in modern times. He also notes the efforts of theorists to introduce still higher numbers, including 11 (his own system) and 13 (Schlesinger's analysis of the ancient Greek scales).

The major advantage to JI is that it attempts to produce consonant intervals that are true to acoustic principles (not mathematical principles, i.e. Pythagorean tuning). The idea that "small number proportions = comparative consonance" is an attractive one, and the intervals within JI are free of the "beats" (wavering in the overall sound, caused by the clash of periodic vibrations) that occur when an interval is tuned slightly away from the small number ratio (Partch, 1949). This beating phenomenon may actually occur from the clashing of the upper partials of harmonic tones, and the dissonance of certain intervals seems to vary on the perception of the individual. In Sethares' terms, this makes the various Just Intonation scales most suited to the timbre of harmonic tones (1999). The relative disadvantage to JI is that the fixed pitch musician can only effectively tune his or her instrument from one starting pitch, and consequently only play in one "key" (Campbell and Greated, 1987). Whether or not this is important to the musician depends on the style of music being played, and those who choose to use JI do so in light of its disadvantages, and may or may not actually be limited by its use.

Tempered Scales

The essence of any *tempered* scale is that the goal of the musician is to play musical material starting on different pitches and keep the same relationship between the intervals as in the original music (*modulation*). This ideal came into Western musical history (although it was actually developed first, independently, in China) during the 17th century, and continues in that tradition to this day (Campbell and Greated, 1987; Partch, 1949). To reconcile fixed-pitch instruments to music that modulates into related keys, Meantone Temperament was developed. The idea behind this tuning was that, for 6 different major keys and 3 minor keys, at least, the thirds would have the same ratio. Although this tuning was popular during a certain period of Western music, there were still keys in which the thirds were badly out of tune (called wolf notes), not to mention the compromise of other intervals remaining inconsistent (Campbell and Greated, 1987; Partch, 1949).

The solution to this problem was the development of 12 -tone *Equal Temperament* (ET). In this tuning, the octave is divided into twelve equally placed degrees, and therefore all of its intervals have the same ratio starting on any pitch. To obtain this scale one has to find the twelfth root of 2 (the octave), and apply that number to each successive equally spaced semitone. The resulting logarithmic formula bears equally complex ratios for the intervals involved in Equal Temperament, yet the goal of modulation to any of the twelve keys is achieved (Campbell and Greated, 1987). It bears noticing that, compared to acoustically true intervals, the tones of any given interval is comparatively dissonant, and the resulting "beats" are to be found in almost any chord. In fact, the common practice in tuning pianos and other fixed pitch instruments is to use the "beating frequency" as a guide for obtaining the correct pitches (Meffen, 1982).

ET is the most commonly used tuning system in the western world today. For this reason, let us look more in depth at the structure of this system. First, the ratios of the intervals in this tuning are so large that a system of naming the pitches by arbitrary letter names, with a few intermediary sharps and flats, has developed. The common standard for tuning to ET is the reference pitch A440 Hz, and here are the pitches (letter names and frequencies), as well as the interval size (in cents) from A440 in that octave:

Table 1.5 - frequencies and intervals in cents between successive pitches in Equal Temperament.

A	A# / Bb	B	C	C# / Db	D	D# / Eb	E	F	F# / Gb	G	G# / Ab
440	466.2	493.9	523.3	554.4	587.3	622.3	659.3	698.5	740	784	830.6
0	100	200	300	400	500	600	700	800	900	1000	1100

<u>Intervals in Equal Temperament</u>

As can be surmised from the above explanation, all of the intervals between pitches in ET can be built off of any pitch without altering their exact size. To name these intervals by their frequency ratios, as in Pythagorean Tuning and JI, would be inefficient because of the large number ratios. For this reason, the intervals in ET have also been given arbitrary names:

Table 1.6 - interval size and conventional nomenclature in Equal Temperament.

# of cents	Interval name
100	minor 2nd (m2)
200	Major 2nd (M2)
300	minor 3rd (m3)
400	Major 3rd (M3)
500	Perfect 4th (P4)
600	diminished 5th (d5)
700	Perfect 5th (P5)
800	minor 6th (m6)
900	Major 6th (M6)
1000	minor 7th (m7)
1100	Major 7th (M7)
1200	Octave (O)

These intervals, unlike those named with frequency ratios, can be added or subtracted together to find their consolidated or reduced interval. For example, a Major 3rd (400 cents) and minor 7th (1000 cents) form a Major 9th (1400 cents).

Other equal temperaments

If the octave can be divided equally into 12 parts, it can be divided equally into any amount of equal intervals, by finding the *n*th root of 2 (where *n* is the number of desired intervals. These scales are also equal temperaments, although not the common one used most often today. Sethares uses the nomenclature *n*-tet to indicate the specific equally tempered scale referred to (1999). Thus 12-tet refers to the scale we have been talking about in the previous sections, and 24-tet would be the same scale with each minor 2nd divided in half. This is also called the *quarter tone* scale, and it is used fairly commonly with musicians familiar with 12-tet, since it only involves finding the mid-point between already learned tones. 36-tet refers to the also popular *sixth tone* scale, so named because it divides the Major second into 6 equal parts. Of course, any number of steps may be used to divide the octave, and experimental electronic composers have tried quite a few. Sethares provides an overview of some of these scales, and a means for finding suitable timbres with which to play them (1999).

Variable Pitch

The ideas about tuning systems described above apply to virtually all pitched musical instruments, mainly as a function of their being able to play ensemble music in tune with each other and have a common frame of reference. However, some musical instruments, like fretless strings, the slide trombone or the human voice, are not bound by a fixed tuning system and are thus able to access pitches in many different tuning systems. These instruments are called *variable pitch* instruments To be sure, the task of producing an exact pitch on these instruments is more demanding than on one where the pitch has been carefully tuned beforehand. Another difficulty in utilizing the capability of variable pitch instruments lies in the capacity of the musician for training their ear to recognize many fine gradations of the pitch spectrum (Snyder, 2000). This is not necessarily an impossible task and an increasing number of contemporary musicians are able to negotiate more than one tuning system.

One other property of variable pitch instruments is worth mentioning in this section. It is the capability of these instruments to produce a continuous pitch motion, or *glissando*. This effect can refer to the smooth transition from one pitch to another without a break in the pitch space or to a wavering path that is held continuously between two or more pitches within the duration of a sound. Fixed pitch instruments are sometimes able to approximate glissandi, whether by "lipping" wind instruments up or down from the fixed pitch, or rapidly moving over the discrete steps of their tuning systems. In any case, glissandi are often inherently related to the pitches and intervals of a tuning system, if only for the fact that one must indicate where to begin (and possibly end) the glissando.

Pitch Capabilities of Acoustic Instruments

In addition to being limited by the capacity to accurately produce specific pitches in different tuning systems, acoustic instruments are also limited to what regions of the frequency spectrum they may negotiate. Due to the fixed length of a string, lip strength of a wind instrumentalist, or number of pieces in a percussionist's setup, they are only capable of producing pitches within certain bands of the frequency spectrum. The difference between the lowest and highest pitch produced by an instrument is known as its *range*. The normal ranges of ten of the eleven instruments recorded for this study (the drumset is omitted, since its component drums and cymbals do not cover a range) are shown in **Figure 1.4**. The range that a musician can access on their instrument varies widely from person to person, and some instrumentalists have developed extended techniques for producing pitches above or below their normal range. The region of pitch above an instrument's normal range is known as *altissimo*, and it should be noted that while some players are able to access specific pitches in this extended range, many simply use altissimo for a piercing high-pitched effect. Similarly, the region below an instrument's normal range is known as *sub-tone*, which may be used for a low, rumbling effect or to produce specific pitches. The means of production that an instrumentalist chooses can also have a great effect on the "pitched" quality of a tone, but we will save discussion of the spectral transformations available to instrumentalists for the section on timbre.

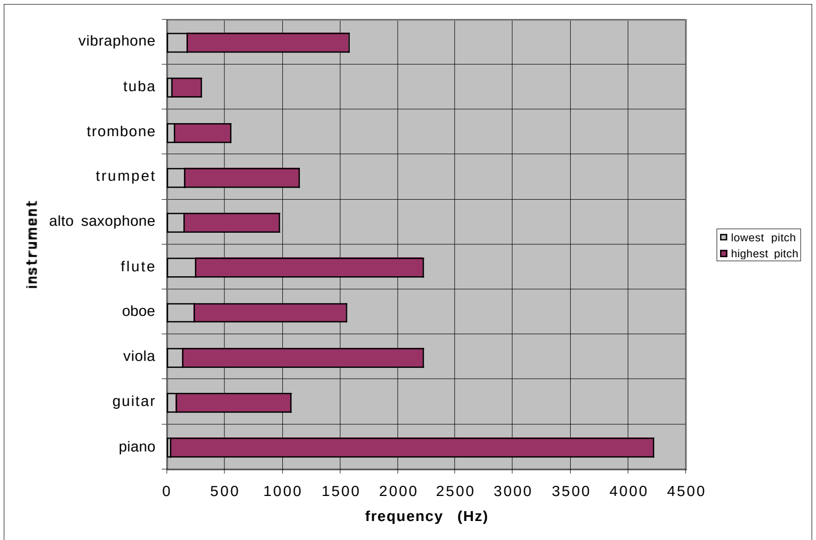

Figure 1.4- normal pitch ranges , in frequency, of ten acoustic instruments.

2. Loudness

Intensity and Sound Power Measurements

Now that we have a grasp on some of the key concepts involved with the sonic parameter of pitch, let us turn to a second parameter: loudness. We saw with pitch that the frequency of a sound wave is an important factor in determining the pitch of the sound. Another aspect of the sound wave is its *amplitude*. If the number of pressure fluctuations in air determines the frequency, the actual amount of pressure determines the amplitude. The loudness level of a sound is related to the pressure level that reaches the eardrums, but this relationship is not as direct as the one between frequency and pitch. When a certain force (such as a fluctuation in air pressure) acts on an object (such as an eardrum), the rate of energy transferred to the object is said to have a certain *intensity*. When a sound is created, the sound wave emitted has a particular *sound power*, (measured in watts, just like a light bulb), which, in acoustic instruments, is typically only a fraction of the energy expended by the player.

Intensity is related to sound power by the equation{$I = \frac{P}{A}$} , where P is the power and A is the unit area. The unit of measurement for intensity is the watt per square meter, which is the energy transferred by a sound wave per second across a unit area. For example, the maximum sound power of a trumpet is 0.3W, so the intensity of the sound wave at its 12cm bell (A= π{R^2}) would be approximately 8.5 Watts per square meter (the information in **Section 2** comes primarily from Campbell and Greated, 1987, unless noted).

As a sound wave spreads out into air from its source, the intensity drops rapidly, as a function of the wave spreading in all directions, like an expanding sphere. If the sound wave is spread equally in all directions, then the source is called *isotropic*. For such a source, the relationship of the intensity of a 1000 Hz tone (a standard acoustical reference) to the distance from the source is I= P(A)/ 4π{R^2} , where A is the area of the window receiving the sound wave (such as an eardrum), and R is the radius of the spherical sound wave (i.e. the distance to the listener). As an example, a 1W isotropic sound wave would have an intensity of 0.08 Watts per square meter at a 1 meter wide window 1 meter distant. Because the intensity depends on {R^2} , the steady increase of distance from the source causes the intensity of the wave to drop at an exponential rate. At two meters our 1W sound wave would only have an intensity of 0.02 Watts per square meter {Wm^{-2}} , at four meters only 0.004, etc.

The Decibel Scale

In order to create a usable scale of loudness from the rather unmanageable decimal point numbers indicated by intensity (which range from 0.01 to 0.000000001), the logarithm of the difference between two intensities (the *intensity ratio*) is used. Just as the logarithm of frequency ratios show us the size of the pitch interval (in cents, see above), so does the logarithm of the intensity ratio show us the size of the loudness interval, called the *decibel* (dB). Thus, if two intensities are separated by the power of ten, the{$\log^{10}(\frac{I1}{I2} = 1$ bel} , subdivided into decibels. For the purpose of setting an absolute logarithmic intensity scale a standard intensity has been chosen: I0= {10^{-12}} (or 0.0000000000001 {Wm^{-2}}). This is just below the lowest intensity that a person with acute hearing can hear a 1000 Hz tone. By comparing the intensity of a given sound with this reference intensity, we can determine the *intensity level* (IL), in decibels, of a sound. Similarly, we can determine the *sound pressure level* (SPL) of a sound by comparing it to the pressure level of a wave of the same intensity as our standard reference (2 X {10^{-5}Pa} (pascal)). This is useful for working with most microphones, which are sensitive to pressure rather than intensity. In a very reverberant room, the IL and SPL of a sound can be several dB different, indicating that what the ear hears and what is picked up by a microphone may not always be the same. In most cases, however, the two terms are interchangeable, and the SPL is also measured in decibels.

Equal Loudness Measurments

The above descriptions of the intensity of a sound wave have been based on the standard reference of a 1000 Hz tone. If the frequency of a sound is changed from this reference, the apparent loudness also changes, even when the intensity of the sound is kept constant. For example, a sound wave with an intensity of 60 dB at 80 Hz will have the same apparent loudness as a 30-dB sound wave that vibrates at 1000 Hz. In order to determine the perceived loudness of a sound in more precise terms, we must look at the *loudness level* (LL), which is measured in equal loudness contours, called *phons* (**Figure 2.1**).

While the phons scale is helpful in determining the perceived loudness of sounds, it has a drawback; the size of the loudness interval is not clearly indicated. It has been discovered experimentally that most people with acute hearing judge the dynamic level of a 1000 Hz tone to drop by one marking when the intensity is decreased by a factor of 10, equivalent to an intensity level decrease of 10 dB. This corresponds to a 10-phon decrease in the loudness level for sounds with different frequencies. Also, in experimental research it has been found that listeners consistently judge the difference between two dynamic level markings to be double the loudness. Thus, since the difference between 50 and 60 phons is double the loudness and yet the values are not doubled, a discrepancy arises. To remedy this problem, the sone scale of loudness has been devised, where 1 sone = 40 phons, and 2 sones is doubly loud (50 phons). It is important to note that both the phons scale and the sone scale are not directly related to actual intensity changes, as is the decibel scale, but to the listener's perception of such changes. To compare the measurements of loudness and pitch, the phons scale quantifies the (perceived) loudness spectrum into discrete units that can be compared as with the frequency ratios in the pitch domain. The sone scale of loudness, based on the logarithm of the phons scale, divides the loudness spectrum into equal ratios (or intervals), much as the cents measurement does to the frequency spectrum. Unlike with cents the usefulness of the sone scale is debatable, given that it is easy to remember that 10 phons = double the loudness, and it is easier to relate to subtle differences in the whole numbers of phons rather than the decimal points of sones.

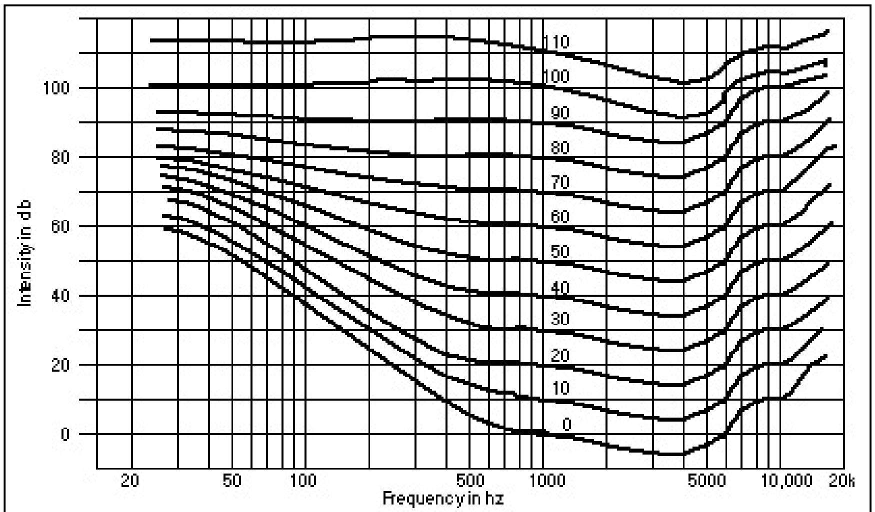

Figure 2.1- equal loudness curves (from Campbell and Greated, 1987).

Dynamic Level Markings

The phons scale is useful in dealing with musical sounds in that it measures our subjective perception of how loud a sound appears to be, in comparison to musical *dynamic level markings* which indicate the learned amount of force for the player to apply. In conventional music notation, loudness is indicated with abbreviations for common Italian terms for different loudness, ranging from pianissimo (which is very soft, marked pp) to piano (p) to mezzo-piano (mp) to mezzo-forte (mf) to forte (f) to fortissimo (which is the loudest, ff). Each of these abbreviations indicates a subjective use of loudness to be played by the instrumentalist, dependant on their experience and strength and also the instrument played. A trumpet playing at full force will obviously sound louder than an acoustic guitar plucking the same pitch as hard as possible, yet both of these sounds would be scored ff. Illustrating that loudness in many instruments is dependent on frequency, **Figure 2.2** shows the varying levels of loudness for instruments playing the same dynamic marking in their respective low, middle and high ranges. The phons scale provides a means for classifying the loudness of any sound by its relation to a tone of standard loudness (1000 Hz sine wave).

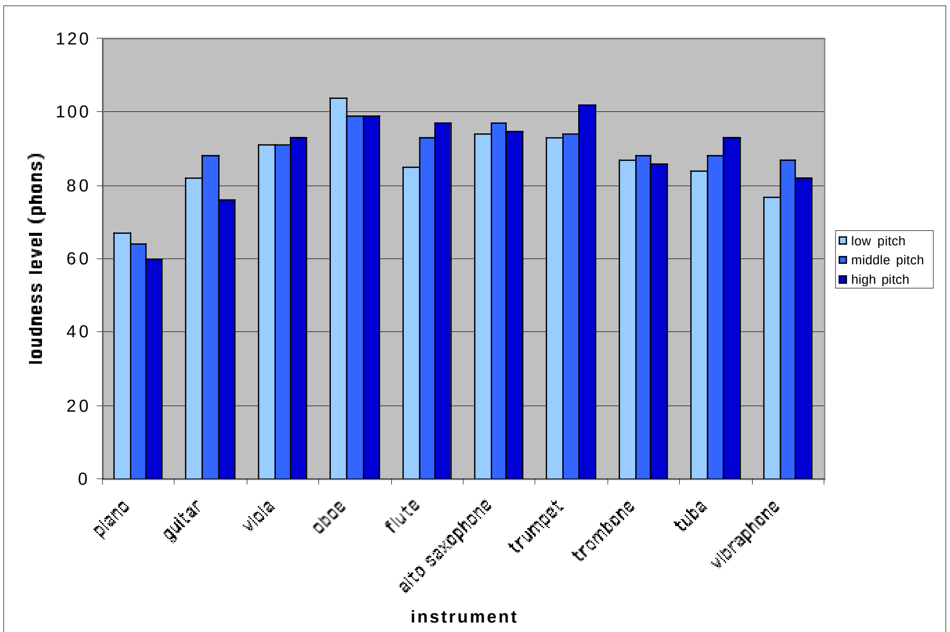

Figure 2.2- loudness levels, in phons, of ten acoustic instruments playing at a mezzo-forte (medium loud) dynamic level in their low, middle and high ranges.

One aspect of dynamic level markings is worth noting. Each marking is an indication of a specific gradation along the loudness dimension of sound. Just as the letter names of pitches in Equal Temperament provide convenient markers for negotiating the frequency scale, the loudness scale is conveniently divided by dynamic level markings, and the decibel and phons scales take this concept to an even more exact level. When working with loudness it could very well be useful to develop rigorous compositional structures in the loudness parameter, as is often done with pitch. In reality, and perhaps due to musicians' lack of awareness, the loudness scale isn't as wide as the frequency spectrum and therefore many fine gradations of loudness are difficult to perceive and work with practically. That said, the effective use of loudness, whether on its own or as subservient to pitch or rhythmic structure, is often quite beneficial to the development of a musical piece.

Another parallel between the pitch and loudness scales is the fact that each can be broken into discrete steps or be negotiated continuously. In the pitch domain, glissandi are relatively rare in common musical practice, due to the limited capacity of many instruments to achieve this effect. A continuous change in loudness is, however, very common in many styles of music since it is relatively easy to execute or simulate on most instruments. The Italian terms crescendo and decrescendo, or swell and diminish, indicate gradient loudness change, and the sounds that they describe are effective in a musical context. Many instruments can produce an effective range of 30 to 40 phons between their lowest and highest sound levels, an equivalent to 8 to 16 times the loudness of the softest tone. In addition, the use of gradient dynamics, usually over a short time scale, can bring an instrumentalist to extend their range a little beyond their normal highest and lowest discrete loudness levels.

<u>Loudness in Complex Tones</u>

Thus far we have been primarily investigating the perception of loudness of pure tones existing at a single frequency. What happens to the perceived loudness of a sound wave when more frequencies are added becomes complicated very quickly. In general, the effect on one frequency of a sound wave by another depends on the distance between them, and whether or not the two tones lie within a *critical band*. If two frequencies have amplitude envelopes that overlap each other on the part of the inner ear called the basilar membrane, then they are said to lie within a critical band, suggesting that they fire the same set of auditory nerves. On the logarithmic frequency scale, the size of the critical band increases with an increase in frequency, much like the size of an octave or other interval increases in Hertz as the frequency increases. Regardless, the effect of two frequencies that lie within a critical band on the perceived loudness of the total sound is that the intensities of the two frequencies are added together. Here it is useful to note that when the intensity of a sound is doubled, this increases the intensity level by 3 dB. Hence, if two pure tones within a critical band both have an IL of 75 dB the resulting total loudness would be 78 dB. When two frequencies are separated by more than a critical band, then the general rule is to add the loudness of both. For example, if our two pure tones each had a loudness level of 75 phons (keeping in mind that doubling the loudness increases the LL by 10 phons), the total loudness would equal 85 phons.

In practical terms, much of the above paragraph is unnecessary for determining the perceived loudness of complex tones. One of the most efficient tools for measuring loudness is the *sound level meter*, which uses a microphone to pick up pressure fluctuations in air. Most sound level meters are equipped with three different weighting networks that tailor the decibel readings to different tasks. The A-weighting setting, for example, roughly estimates the equal loudness contour of 40 phons, and is the most standard and efficient means of quickly determining the perceived loudness of a sound. In order to measure the actual sound pressure level with as little weighting as possible, the C weighting scale should be used. Obviously, these readings can then be converted to phons by referring to the equal loudness contours.

Loudness in Acoustic Instruments

Comparing the acoustic theory of loudness with the common musical practice of indicating relative force used by the musician, one finds that there is a great gap between the classification of a sound's loudness by its amplitude and the idea of its loudness in the mind of the musician. In order to bridge this divide, a long-term goal would be for musicians and composers to add sound level meters to their arsenal of tuners, metronomes, etc. While consistent practice with such a device could replace dynamic level markings with the more precise terminology of decibels and phons, such a solution isn't easily undertaken. For today's musician a rough knowledge of the range of loudness available on the different instrument types will go a long way. With this knowledge comes not only a more precise understanding of what loudness is and how it relates to musical performance, but also a means of plotting future compositions of sounds with precise amplitudes and structured sequences of loudness integrated with the other parameters of sound.

3. Timbre

We have seen that the frequency components of a sound influence its perceived pitch, yet the makeup of those components also influences a sound's *timbre*. According to the American Standards Association, "timbre is that attribute of auditory sensation in terms of which a listener can judge that two sounds similarly presented and having the same loudness and pitch are dissimilar" (quoted in Malloch, 2000). This official definition of timbre has been subject to intense criticism in the past quarter-century, and many recent studies on timbre analysis point out that the definition is too broad to be useful (see Erickson, 1975: Deutsch, 1982: Slawson, 1985; and Sethares, 1999). A wide range of phenomenon fall within this category, but of all of the factors that determine a sound's timbre, two major parameters contribute the most. One is the *spectrum*, or spectral envelope, of the sound, which is a means of representing the average frequencies present in a sound and their relative amplitudes. Spectral analysis, then, looks at the distribution of the frequency components and attempts to order these on various scales (bright to dark, sharp to dull, by comparison to speech vowels, etc.). These characteristics usually change over the course of a sound's duration, and for this reason the *envelope* is the second major determinant of timbre. Many musical effects, such as vibrato, trills, tonguing, and slurring can only be recognized when the analysis of the envelope is added to that of the spectrum. Also, it is said that certain characteristic portions of the envelope (particularly the beginning) help the ear to determine what instrument is playing a given sound.

The relationship of the spectrum and envelope to our perception of a sound is complex, and the study of this field is still new and developing in comparison with the analysis of pitch or loudness. However, detailed scrutiny of these two factors gives us a fairly good method for comparing the differences between sounds of equal pitch and loudness, and thus provides a means for increasing our understanding of timbre as a major sonic parameter. The field of timbral analysis has benefited greatly in the past quarter-century by a vast amount of research on the properties of musical spectra, and especially by the development of a multitude of theories on how to characterize those spectra (Malloch, 2000). While many of these theories are useful to those interested in the musical applications of timbre, the field is as yet to young to produce a definitive, all encompassing set of rules for spectral analysis. Therefore, to gain a clearer understanding of the multi-dimensional parameter of a spectrum it is necessary to combine the analysis of a number of different acoustical measurements related to the spectrum of a sound, which we will look at below. First let us examine some related concepts having to do with the analysis of a sound's frequency components.

<u>Waveforms and spectral envelopes</u>

As we saw in the section on pitch, the waveform of a sine tone looks like a gently rolling curve, with successive crests relatively compressed or spaced out, indicating the frequency (cycles per second). Complex tones can be thought of as a collection of sine waves at different frequencies, and the effect of this phenomenon on the waveform is that the sine waves are added together, causing a waveform with "bumps" and "wiggles" between the crests (**Fig. 3.1**). Here is where we find the first indication of the spectral characteristics of a sound; in general, the more bumps and wiggles in a waveform, the more frequency components it has, and thus a "richer" or "brighter" timbre

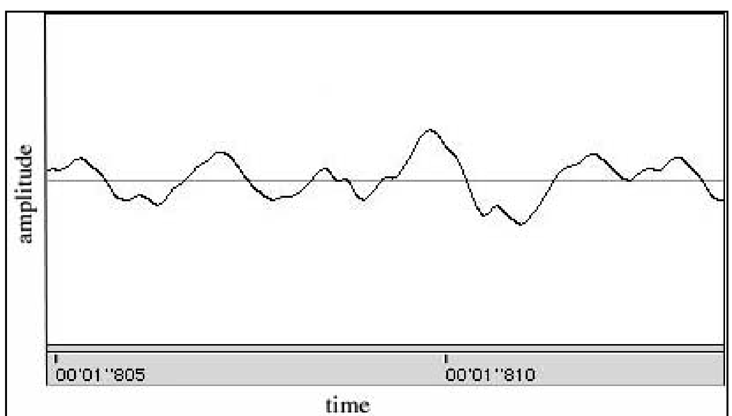

Figure 3.1- waveform of a viola sound.

(Matthews, 1999). It should be stressed here that we are describing the waveform at the micro-level of each successive crest and trough; the overall variation in amplitude of sound is an amplitude envelope, described later. Visual waveform analysis is not capable of much more than a relative bright/ dark indication, however, because of the *phase* of the different frequencies. Each of these frequencies is liable to begin its cycle at a different point in time every time the same pitch is played on an instrument, and this change is called the phase difference (Campbell and Greated, 1987). What this means to the waveform is that its bumps and wiggles are likely to be in different places in relation to the crest, and therefore to get a better picture of how these frequency components are related we must turn to another method. Aurally, however, it has been determined that phase differences in sound with like spectra play little or no difference to our perception of those two sounds (except in very low frequencies) as being of the same type (Pierce, 1999).

In fact, there are many methods of spectral analysis, but the most popular (and the methods that we will be concerned with here) are a family of techniques known as Fourier analysis. These translate the information in the waveform into individual frequency components, each with its own amplitude and phase (Roads, 1996). The resulting visual representation can take a variety of forms, but one common one is the *spectral envelope*, which is a two dimensional graph with the frequency spectrum on the horizontal axis and amplitude on the vertical axis (**Fig. 3.2**).

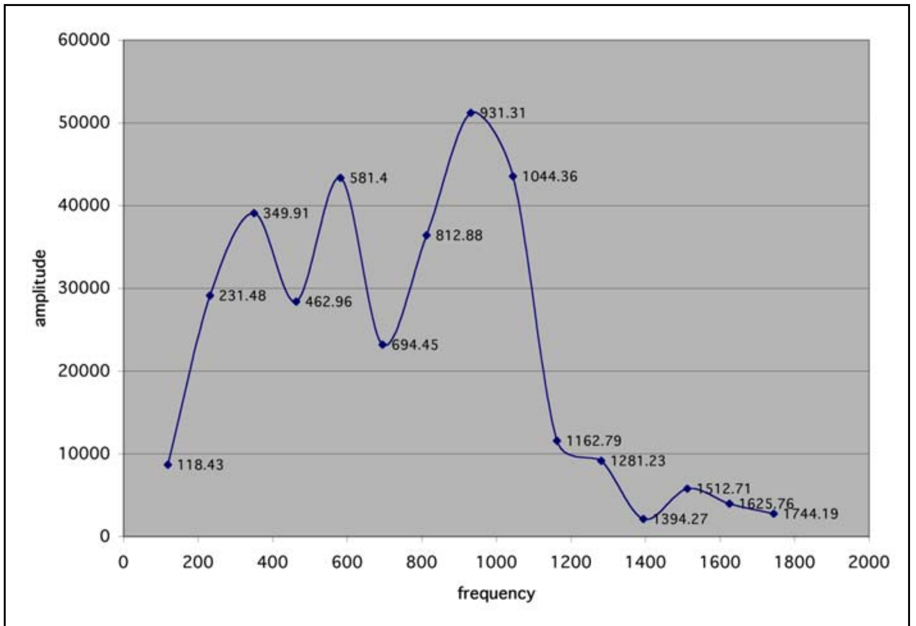

Figure 3.2- spectral envelope of a trombone sound.

The peak amplitudes of the frequencies in a spectral envelope are connected to form a smooth curve indicating the spectral peaks and valleys of a given sound.

Resonance

One question that arises when viewing the spectral envelope of a sound is why do specific frequency components have greater amplitude than others do? Most vibrating systems (including musical instruments) have inherent in their design the fact that some frequencies will be reinforced by the different properties of the system, such as the materials, shape, any open cavities, etc. When a specific frequency is reinforced by the structure of a vibrating system the system is said to have a *resonance* at that frequency. A resonance curve shows the defining characteristics of a resonance: the *center frequency* and *bandwidth*. The center frequency is the frequency at which the curve has its peak amplitude. The bandwidth describes how steep the curve is, and is defined by the frequency difference between two points of the curve on either side of the center frequency that are 3 dB lower than that peak. A narrow bandwidth indicates that the resonance curve has a sharp peak, and thus only a small frequency range is reinforced by that resonance. Such a curve is said to have a high *Q, or quality factor*. Conversely, a wide bandwidth, dull peak (low Q) will reinforce the center frequency and many around it (Slawson, 1985). Each major peak in the spectral envelope indicates a resonance in the sound source, and thus gives us clues about the timbre of the instrument creating the sound.

Formants

When analyzing spectral envelopes of acoustic sounds, one common phenomenon that occurs is that specific frequencies do not often stand out alone in a particular region of the pitch spectrum, but if one frequency shows a peak, adjacent frequencies also have relatively large amplitudes. When an entire region of the pitch spectrum within a spectral envelope is raised, this region is called a *formant*. These are essentially resonances whose Q factor is sufficiently broad as to affect a number of overtones contained in a sound. The study of human speech has provided a wealth of research into the properties of formants since their location in the frequency spectrum is the determining characteristic of the vowels that we create with our voices. When we create vowels in our everyday speech, what we are essentially doing is changing the shape of our oral cavity to produce resonances at particular regions of the pitch spectrum, and this causes formants to show in the resulting spectral envelopes. Obviously one can sound a vowel, say "ah", at a number of different pitches and it retains the "ah" quality regardless (**Fig. 3.3**). We could therefore say that all cases of the vowel "ah" show a similarity in timbre. In this case, what unifies the spectral envelopes of each of the "ah" sounds is that the formants are in the same place on the pitch spectrum as all of the others. In **Figure 3.3**, there is a strong formant in the 600-700 Hz region for both pitches, showing how formants that remain invariant produce the same timbre, or "sound color", according to Slawson (1986).

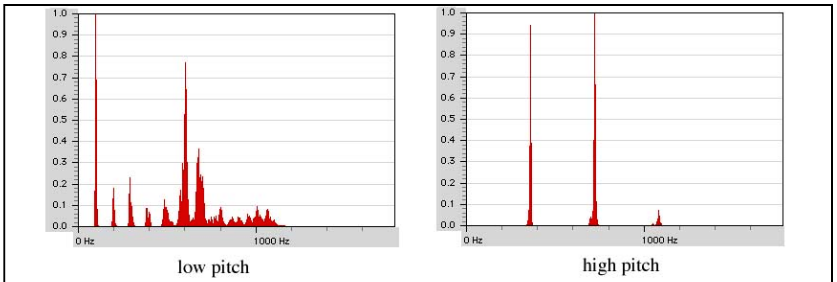

Figure 3.3- two examples of the vowel "ah", illustrating a formant region around 600-700 Hz.

The reason that voices are able to produce this phenomenon is that the sound's source (the vibrating vocal cords) is *weakly coupled* to its filter (the oral cavity). We are therefore able to retain the correct mouth and tongue positions for any given vowel, and the arbitrary pitch produced by our vocal cords is filtered through the same resonances, causing the characteristic formants in the spectral envelope. In most other acoustic musical instruments, however, the source and filter are *strongly coupled*. In this case, the formants change as the length of tubing or string is lengthened or shortened, causing new resonance peaks in the spectral envelope. There may also be specific resonances that are found in certain parts of an instrument that have an effect on the spectral characteristics across its pitch range. It is known that the shape and type of wood used on violin bodies is carefully crafted to accentuate certain tones in its range.

The idea of relating the formant regions of a spectral envelope to the resonances of human vowel production creates an attractive analogy that any sound's timbre may have a corollary with a particular vowel. This type of model would appeal to many musicians who seek an easy means of relating to what might be a somewhat arbitrary-looking set of number measurements. One can perhaps think of numerous examples of how an instrument's timbre is related to vowels in everyday conversation and music (an example can be found in jazz and pop music "scatting", where a particular instrument is imitated with a suggestive vowel type). Unfortunately, scientific research still has yet to fully understand the complex phenomenon of vowel production, and one of the problems that interferes with relating vowels and other sounds has to do with the process of *normalization*. Setting aside the differences between strongly coupled acoustic instruments and weakly coupled voices, we find that even among voices the size of the vocal cords and oral cavity can cause drastic differences in the location of formant regions of spectral envelopes for the exact same vowel. This is known as normalization and explains the fact that a child can speak an "ah" sound with much higher-pitched formants than an adult male (even on the same pitch), yet the two sounds have a similar timbral quality. Researchers have been able to analyze and predict differences among voices given the size and shape of the oral cavity for an individual, but have yet to reliably explain how strongly coupled instruments might also be "normalized" to associate the particular spectral envelope to a vowel type (Slawson, 1986).

One reason for the confusing nature of a sound's perceived relation to vowels is that even sine tones are perceived as being similar to different vowels in different parts of the frequency spectrum. A low frequency sine wave, for example, resembles an "oo" or "oh" sound, while a high pitch sine wave more closely resembles "ih" or "ee" (Cogan, 1984). The importance of this phenomenon is most marked when a low-pitched sound with a wide spectral envelope has many prominent upper partials, thereby confusing our normal expectation of a sound's similarity to a vowel.

One of the attractions of the vowel formant/ spectral envelope model of timbre is that a multi-dimensional space can be formed that is capable of being subjected to complex operations and permutations, much like our current system of negotiating the pitch parameter. Slawson's book Sound Color (1986), provides many examples of how this space can be mined in the composition of interesting musical works involving rigorous timbre change concepts that agree with our perception of the sounds. We will investigate those processes more in Part Two of this essay. There are other timbral measurements, however, that can reduce the information available from spectral envelopes into useable scales of timbre differentiation. As Malloch points out, it may be advantageous to combine these measurements to form more complete representations of a sound's timbre, and this will be the approach taken here (2000).

The Tristimulus Method

One other useful system of timbral analysis is called the *tristimulus method*, which originally was set up to show how certain regions of a harmonic spectrum of are more accentuated than others. In the original method, the relative amplitudes of three regions of the overtone series (the fundamental, partials 2-4, and partials 5 and up) are reduced to single percentages of the total envelope amplitude. The second and third regions are then plotted as a single point on a graph. Although there are only two numbers represented, information about the third (the strength of the fundamental) can be inferred by the position of the point on the graph. For example, a sound with a strong fundamental (and few upper overtones) would be found near the origin point of the graph, and as partials 2-4 increase in amplitude, the corresponding point would move towards the right. An increase in higher partials would cause the point to move vertically up the graph.

This method definitely does create a useful model of timbre perception for harmonic tones, but its usefulness is limited to just such sounds, in much the same way that the vowel/ formant model is best suited to analyzing vocal tones. However, Malloch has proposed a variation of the tristimulus method, called *Loudness Distribution Analysis* (LDA), that is useful not only in the analysis of non-harmonic tones, but in analyzing ensemble passages of music as well (2000). This method also divides the spectral envelope into three distinct sections, only this time the key criteria lies in determining the 1/3rd octave frequency band that has the highest amplitude. This measurement replaces the fundamental in the tristimulus method, and it is also useful to determine the frequency position of that 1/3rd octave band, known as the *timbral pitch*. Another measurement in Loudness Distribution Analysis is *timbral width*, which is calculated by determining the total loudness of the regions above and below, but not including, the loudest 1/3rd octave band. This is perhaps the most important measurement in LDA, as it scales timbre along a range from focused to diffuse, which can agree with our perception of a sound (see **Fig. 3.4**). *Timbral weight* is measured by subtracting the amplitude of the region below the loudest band from the region above that same band, the resulting number indicating the relative shape of the envelope. High numbers would indicate that upper partials are prominent, and negative numbers indicate that most of the energy in the spectrum lies below the loudest band.

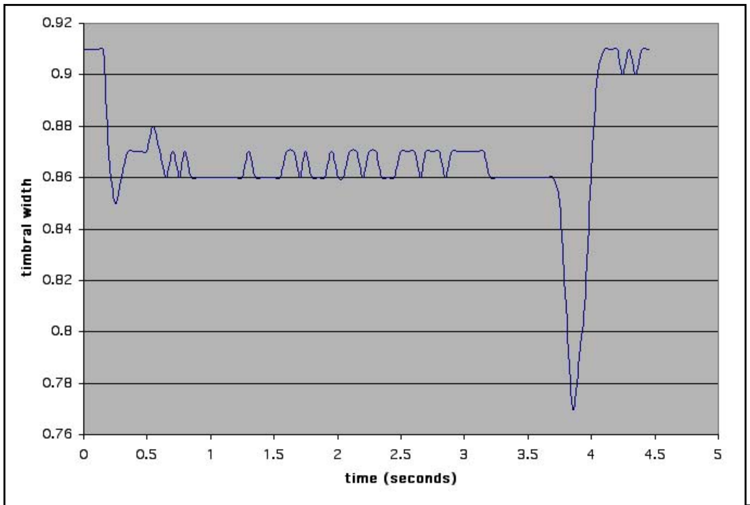

Figure 3.4- timbral width of a mid-range trumpet tone, over the span of its duration.

<u>Sharpness</u>

While Loudness Distribution Analysis does give the viewer an easy means for analyzing timbre that is consistent with our perception of the sounds viewed, even the creator of this method suggests using LDA as one part of a holistic approach to timbre analysis (Malloch, 2000). There are other measurements that can be derived from the properties of a sound's spectral envelope that also give a clear basis for scaling a group of timbres according to our perception of them. One of these is *sharpness*, which analyses the frequency of the *centroid* of the spectral envelope, which is the point at which the envelope divides into two equal sections. If the centroid and the loudest 1/3rd octave band happen to fall in the same place on the frequency spectrum, then the sound's timbral weight would be 0, and the centroid would also be the same as the timbral pitch.

It is important to note that, when measuring timbres using Loudness Distribution Analysis or sharpness scales, a number of different-sounding timbres can share the same measurements. Zwicker and Fastl point out that, for sharpness, it matters little whether how many frequencies fall within the regions of an envelope, "even when a critical-band noise is used for comparison" (1990). What does matter, and this is the case for LDA as well, is how the amplitudes of any frequencies are distributed. In practical terms, this means that a harmonic instrumental tone may have the same sharpness as an electronic noise, yet these two sounds have very different apparent timbres. On the other hand, one would be able to distinguish clear similarities in the sound "color" between disparate sounds of like sharpness, by the perception of whether upper or lower frequencies of the sounds are relatively prominent or not. In the case of timbral width, whether or not the sounds are focussed or diffuse would also be apparent to the ear. **Figure 3.5** shows sharpness over time for low and high pitches played by the flute. Notice that the higher pitch has a fairly constant sharpness, probably due to its strong fundamental and lack of overtones, while the lower pitch has a wavering sharpness due to its great number of upper partials.

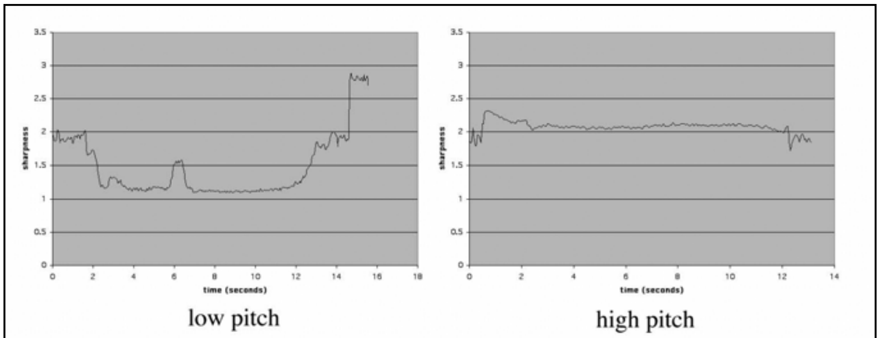

Figure 3.5- sharpness of two flute tones (low and high pitches) over time.

Roughness and Harmonicity

In addition to analyzing the spectral envelope and how the balance of frequencies relate to our perception of timbre, there are also measurements that determine whether any individual frequencies within the envelope interfere with each other, thereby affecting our perception of the sound as a whole. *Roughness*, sometimes called *acoustic dissonance*, is a measurement that calculates any beating between frequencies that lie within a critical bandwidth. A sound with a high roughness value would be characterized as "fluctuating" or "unstable." Different acousticians have developed various formulas for calculating the roughness (or dissonance) of a spectrum in an attempt to more closely model human perception, but it has yet to be determined which of these formulas are most accurate (Cabrera, 2000).

Other types of measurement can calculate how closely the frequency components of a specific spectrum resemble a theoretically ideal harmonic series. These calculations determine a sound's degree of *tonalness*, or *harmonicity* (Cabrera, 2000). Again we see a correlation between the pitch and timbre parameters, for using measures of tonalness can be helpful in determining the clarity of pitch for a sound, as we saw in the discussion of the perceived pitch spectrum above. The relative ambiguity of pitch can also be a factor in our perception of a sound's timbre, and this property is related to the reinforcement of the harmonic series by a sound's overtones.

From the above discussion of spectral analysis, it is clear that there are a number of methods of measuring the timbral quality of different sounds according to their spectra. Each of these is useful in determining how different parts of the spectral envelope and its components affect our perception of timbre. Since timbre truly is a multi-dimensional space, then it is useful to observe how these different factors combine in our perception of both single sounds and musical passages (Malloch, 2000). As yet, however, there has not been a theory propounded on how this might be most effectively achieved. Work by Cogan may offer a means of combining spectral and other measurements into a single value representing the timbre parameter, and we will look at this in more depth later (see **Section 5**). For now it is enough to note that the above methods of spectral analysis have important implications for musicians, who most often base their descriptions of timbre on the source of a sound, rather than its perceptual attributes. If it is possible to train oneself to hear differences between sounds along the spectral measurements described above, and this is most likely, then they could prove to be powerful tools for ear training, especially with those interested in music that emphasizes timbre change.

Instrumental Timbre

Now that we know what aspects of a sound's spectrum are perceptually meaningful in timbre analysis, it would be useful to observe how instrumentalists in the real world negotiate timbre changes on their instruments. While it is true that many people use the word timbre to describe the quality of overall sound of a particular instrument, in reality there are often a wide variety of timbres that can be produced on a single instrument, even without electrical manipulation. However, many composers use the alternation of instrumental timbre as a means of creating change in the timbre parameter in a linear fashion, as with pitch and loudness. **Figure 3.6** illustrates how the same pitch on different instruments produces

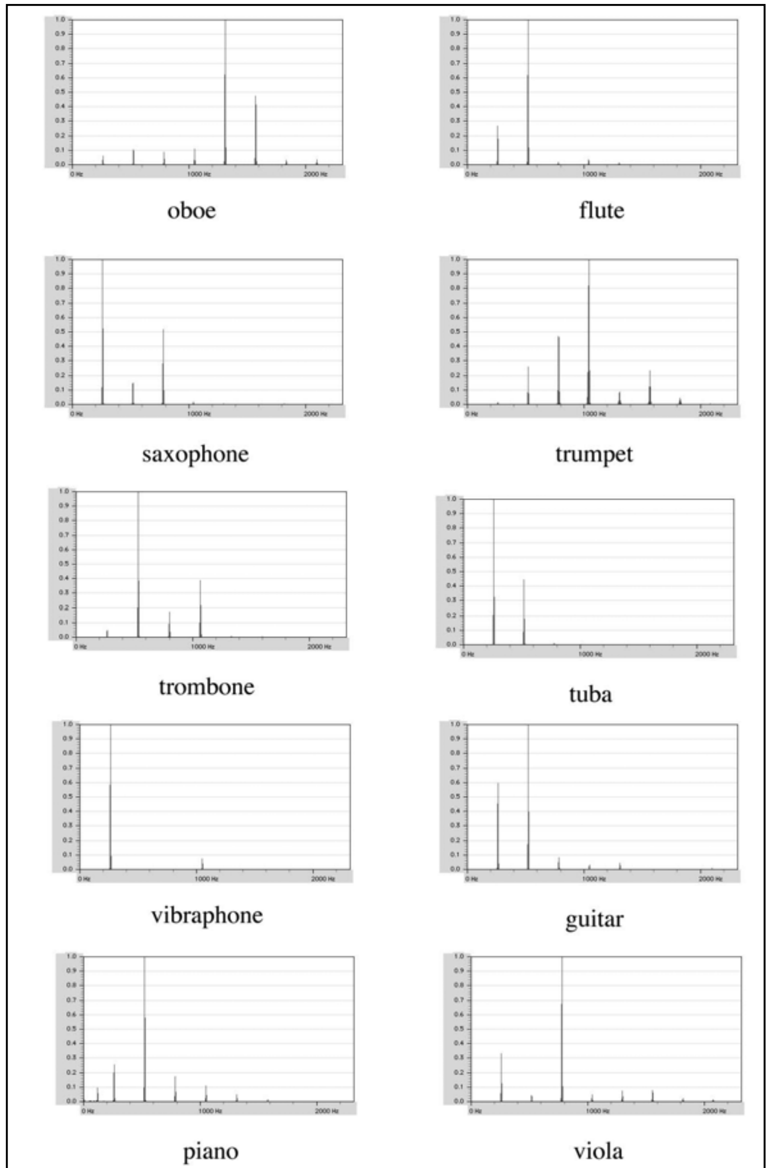

Figure 3.6- spectra of ten acoustic instruments playing the pitch C4 at a medium dynamic level marking.

many different spectra. Many of the techniques used to vary the timbre of a single instrument (and therefore the sounds that these techniques produce) are now increasingly common in many styles of music, although this is a fairly recent development (Stone, 1980). For this reason, while many of the "extended" techniques of acoustic instruments are common, it is not safe to assume that all musicians can produce these sounds on their instruments.

On a number of wind and string instruments, it is possible to alter the timbre of the instruments in a continuous fashion (sometimes referred to as spectral glide), as well as by discrete steps Erickson, 1975). For wind instruments, this usually involves tightening or restricting the airstream entering the instrument in a highly controlled manner, thereby accentuating upper overtones in the spectrum. Bowed string instruments can also change timbre in a continuous fashion by moving the bow closer to or further away from the bridge of the instrument, the closer to the bridge the more upper overtones are accentuated.

<u>Rapid Re-articulation</u>

For those instruments that aren't able to alter the timbre of a sound once it is set into motion, like plucked and struck strings and percussion, there is one technique available that allows one to affect continuous timbre change: rapid re-articulation. *Tremolo* and *fluttertonguing* involve the rapid re-articulation of a single tone, and these numerous attack points are treated perceptually as one sound with its own duration (Erickson, 1975). The cognitive process that causes this to occur is known as *event fusion* (Snyder, 2000). A *trill* is similar to a tremolo except that it is a rapid alternation of two separate tones, forming the basis for fusion into a more complex timbre. As the musician is able to sustain a tone through rapid re-articulation they may also be able to create a gradient timbre change.

While rapid re-articulation makes it possible to manipulate a sound's timbre within it's duration, it also alters a sound's spectrum to apply a rapid re-articulation in the first place. We will examine the different aspects of a sound's temporal envelope more below, for now it is enough to note that the beginning of a sound, or attack, normally has many more inharmonic partials than in the remainder of a sound's duration. When these attacks are rapidly re-articulated, the inharmonic partials play a greater role in the makeup of a sound's spectrum that is, after all, an average of the frequencies in a sound. Therefore, the spectrum shows the effect of this noise. **Figure 3.7** shows the effect of tremolo bowing on the normally harmonic spectrum of a viola tone. There are also other techniques that musicians can use to create sounds with more of what Erickson calls "rustle noise", such as blowing just breath through a wind instrument, or crackling the hair of a bow (1975).

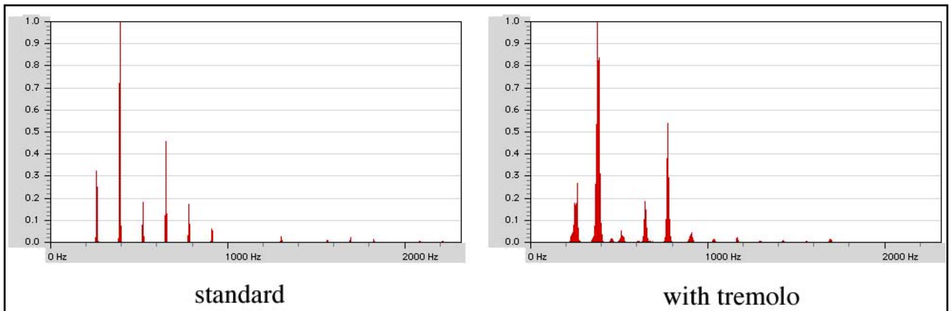

Figure 3.7- the effect of tremolo bowing on the spectrum of a low-pitched viola tone

<u>Multi-phonics</u>

One other set of techniques is common in modern instrumentalism for affecting change in the spectrum of a sound, and these are known as multi-phonics, literally "multiple sounds." There are many different techniques of creating multi-phonics on various instruments, but these can be grouped into two families. Before we look at these, though, a little background; on many instruments, more than one pitch can be created in each physical position, often with some overlap, and musicians sometimes use these *alternate positions* to create a timbre change in their music. A pitch that sounds rich and full when played in a lower position of one string may sound thin or dull when played in a higher position on a lower-tuned string, and musicians often use both types of timbre depending on the context of the musical situation. If it is possible to create more than one tone in a position, then it often is possible to alter mouth or bow pressure so that the frequencies in both tones (the fundamental and overtones) a sound at the same time. This type of multi-phonic, called a *split tone*, occurs when a musician deliberately accesses more than one set of resonances in the same physical position on their instrument. These sounds can range from being very noisy to very harmonic, depending on what resonances are accessed and how their frequencies beat against or boost each other.

Another way to create a multi-phonic sound on an acoustic instrument is to add a separate sound source to the original. This is similar to a split tone, but in this case the secondary sound source comes from a different vibrating system. An example would be when a wind instrumentalist sings through their instrument while playing simultaneously. In an extreme case of perceptual fusion, it is possible to hear two simultaneously sounding tones from different sources (a *chord*) as a single timbre, if certain conditions of timing and loudness are just right. In fact, as Erickson points out, there has been increasing confusion over what constitutes a chord, timbre or pitch (which has an inherent timbre), because of the new developments in pitch and timbre combination that has occurred in acoustic and electronic music since the beginning of the 20th century (1975).

It is beyond the scope of this essay to attempt to analyze all of the myriad combinations of pitches that can be sounded on a given instrument as chords, but one phenomenon that occurs when two or more tones is sounded is relevant to any type of multi-phonic. Often the illusion of a third tone (or fourth or more) can be created when multiple tones are sounded on the same instrument. This is known as a *combination tone*, and they are strongest when the actually sounding tones form an incomplete harmonic series. A pitch interval such as 4:3 strongly implies the 2:1 and fundamental below it and for this reason the cognitive processes in our auditory brain fill in this missing information (Campbell and Greated, 1986).

It is easy to see how analysis of the overlap of frequencies in a multi-phonic becomes complex quickly, yet it is surprising that as yet little or no acoustical research has been done on these and other extended techniques. Most theories of timbre in acoustics are either sufficiently general that any sound can be applied or they deal only with the differences in timbre that occurs with loudness change on an instrument played with standard technique. Since the advent of computer music in the last quarter century, this lacuna may indeed be remedied, as more and more complex models of timbre are developed into musical works and become standard.

Envelope

Since we now have an understanding of how the average frequency content (the spectrum) of a sound influences our perception of it, we can investigate how these frequencies and their relative amplitudes change over time. The change in all of the sonic parameters over time is known as the *envelope* of a sound. Any individual parameter or measurement thereof can also be represented by its own envelope by observing its behavior over time. It is therefore possible to speak of the sharpness envelope of a sound or the amplitude and frequency envelopes of each individual overtone in a sound. To complicate matters, issues relating to both frequency and amplitude (again, over time) both play important role in our perception and differentiation of timbre. Of these numerous types of envelope, the one most characteristically referred to is the *amplitude envelope*, which is the behavior of the total loudness of a sound over the course of its duration.

The amplitude envelope usually is divided into 3 sections, each playing a role in our perception of a sound. The *attack* (or articulation) is a brief time at the beginning of a sound where noisy *onset transients* from the creation of a sound decay into the *steady state* (or sustain) portion of the envelope, which is often averaged over time to determine the spectrum. Lastly, the *release* portion of the envelope occurs when the energy used in creating the sound is stopped and the amplitudes of any frequency characteristics decay to the end of the sound. These terms come from the electronic music studio, where ADSR envelopes are imposed on a waveform to create a more "natural" timbre. The variety of envelopes found in acoustic instruments is large, but it should be noted that certain instrument types, plucked or struck strings and percussion, for example, have no steady state in their envelopes (see **Fig. 3.8**) due to their means of sound production. Although the spectrum of a sound is a key determinant in recognizing different timbres, it has been proven experimentally that the differences between timbres of sounds who have had their attack section removed are ambiguous and harder to differentiate

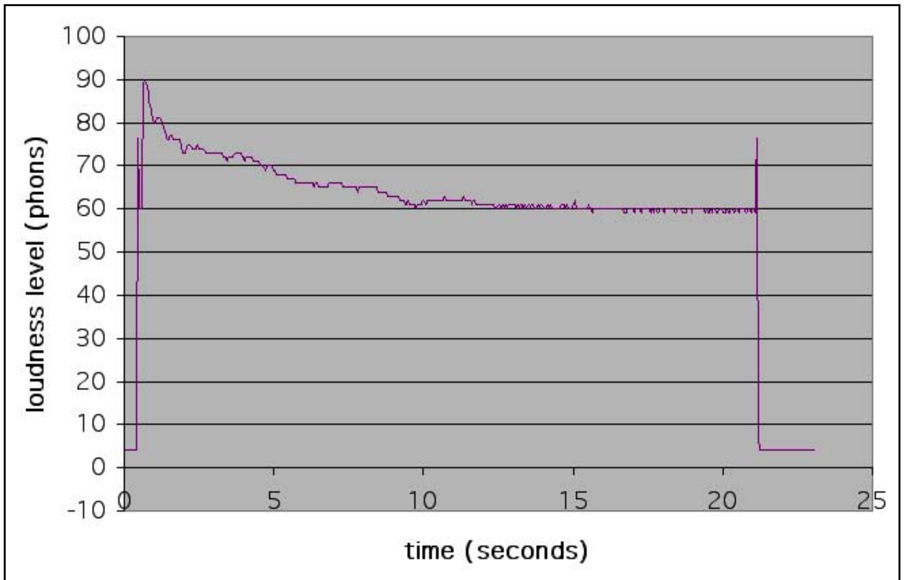

Figure 3.8- amplitude envelope of a mid-range piano tone.

than when the entire sound is presented (Deutsch, 1982).

As we saw with spectral manipulation, musicians have developed many resources for altering a sound's envelope within musical contexts. In a way these techniques are common, in that these manipulations of the micro-timing of an envelope are what musicians use to create expressive changes in both written and improvised music. Manipulations of the attack portion of the amplitude envelope are common, and we saw above how rapid re-articulations such as fluttertongue and trills are used to affect spectral change, yet musicians frequently vary the attacks of single sounds as well. When a musician plays in a standard manner (according to Western classical technique), the envelope generally has a fairly strong attack, steady sustain, and the decay and release occur slightly before the full duration might end and the next begins. *Legato* playing, by contrast, tends to aim for the full duration and less accentuated attacks, while *staccato* playing is characterized by a sharp attack and a clipped sustain/ release, leaving more space between sounds.

Vibrato

There is one more mode of playing that we will look at in this section that musicians commonly use to vary the envelope of a sound. *Vibrato* is a wavering quality in a sound caused by micro-fine alternation between small amplitude and frequency changes over the course of a sound's duration. Again this technique is most common with instruments that have control over the timbre parameter during a sound's duration, like wind and bowed instruments, although vibrato can be used on some plucked or struck instruments. One percussion instrument studied for this project, the vibraphone, has a motorized mechanism that causes the amplitude of its sounds to fluctuate at a speed set by a rotary dial. Because the techniques of producing vibrato are different for each type of instrument, certain instruments utilize vibrato that emphasizes frequency variation, and others emphasize amplitude change. Usually, though, a sound with vibrato will cause fluctuations in both the frequency and amplitude domains, regardless of the musician's intent (Campbell and Greated, 1986). Since in the electronic music studio the terms *frequency modulation* (FM) and *amplitude modulation* (AM) are used to indicate the wavering effect used in place of natural vibrato, it is convenient to apply these terms to the techniques of acoustic instrumentalists. The gentle shaking of a wind instrumentalist's embrochure (mouth placement) would generate FM, for example, while the gentle fluctuation of power in their diaphragm would be analogous to AM.

With many of the above methods used to manipulate the envelope of a sound, it is also possible to vary the speed at which some of the effects take place, usually in a continuous fashion. This micro-level alteration of the speed of events within a sound also plays into the following discussion of duration. Vibrato, tremolo and trilling can be slow or fast or any combination thereof, and the choice of this is often related to a particular player or style (musical context). Even the beating of simultaneous frequencies that occurs during the spectral transformation of a multi-phonic can be sped up or slowed down by the micro-tuning options available on some instruments. Given all of the timbre manipulation techniques available to instrumentalists, especially in light of the timbre changes that naturally occur with pitch and loudness change, the possibilities for composition of effective music based on timbre seem almost endless. One of the biggest obstacles for musicians looking to explore this field has been the lack of cohesive standards of terminology and measurement of the many aspects of timbre. Hopefully the above section consolidates some of this information, setting the stage for a discussion in Part Two of how forms of music utilizing timbre manipulations can be created.

4. Time

We have so far investigated the means of analyzing the sonic parameters of pitch, loudness and timbre. We also saw that the micro-fine timing of change in the parameters of a sound (the envelope) can play an important role in musical structure and interpretation. In addition, the timing of events between individual sounds in a musical work also plays an important role in musical structure and nuance. For this reason we can speak of time in general as the fourth major parameter of sound. The total length of time that it takes a sound to complete an envelope is known as its *duration*, which is a unit of measurement (based on the second hand of a clock) in the time parameter. The physical limitations of playing acoustic instruments, such as lung capacity in wind instrumentalists or the length of a viola bow, play an important role in how long a sound can be sustained. In addition, for some instruments duration is also dependent on frequency (see **Fig. 4.1**). This may be due to the time it takes for a plucked sound to decay fully, for example, or to the fact that lower pitches on wind instruments take more air to sustain. We will examine the timing and rates

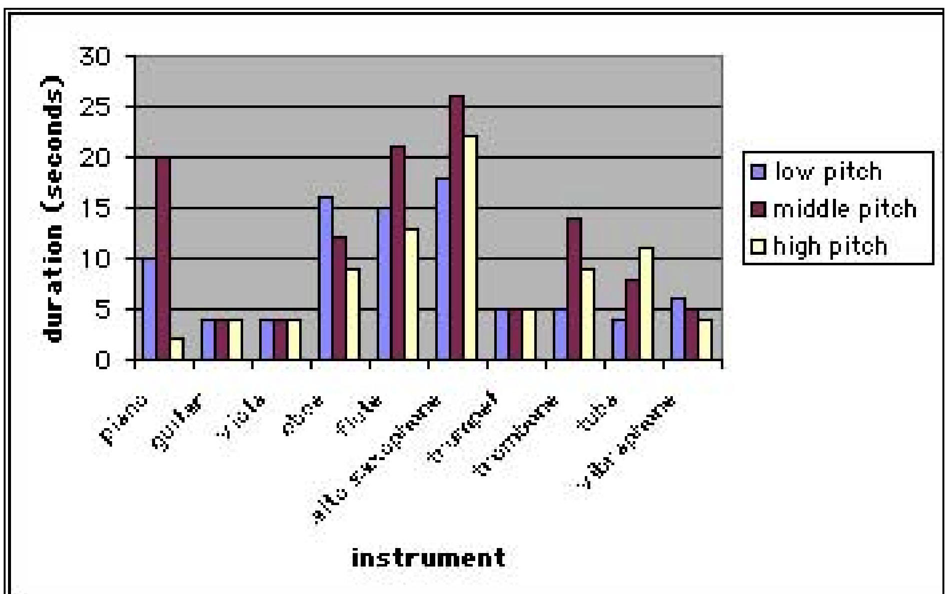

Figure 4.1- maximum duration for low, middle and high pitches in the ranges of ten instruments.

of change within a sound's duration in more detail in **Section 5**. The structure of timing and duration in a piece is known as *rhythm*, and there are basic phenomenon associated with rhythm that we will look into in this section, as a transition into the discussion of rhythmic processes used in composition (**Section 8**).

<u>Rhythm</u>

Due to the fact that there are few theories of duration and rhythm that attempt to explain the fundamental characteristics of this parameter, this section will necessarily begin such an attempt. There are a wide variety of rhythmic structures in music, but two main types are most important for our discussion: metric and non-metric. *Non-metric rhythm,* or *event-based time*, essentially proceeds from one sound event to another, and for our purposes could be described by the number of articulations or events per second. *Meter*, on the other hand, is a means of ordering rhythms according to a common *pulse*, or series of regularly recurring articulations, whether stated or implied (Snyder, 2000). With metric rhythm, the absolute time scale of the second hand of a clock may be less important than the speed of the common pulse of the music, or *tempo*, which can vary across a wide range. Musicians commonly measure tempo in beats (pulses) per minute. Schafer proposes that this tempo scale be reduced to pulses per second in order to align it with the frequency scale, a convention utilized in this essay (1969). A *beat* is defined as an individual articulation in a pulse, so that we might talk of a pulse that is three beats long, or 8 beats, or 57 beats, etc. (the primary source for the discussion of rhythm in this section is Snyder, 2000, unless otherwise indicated).

If we determine the tempo of a section or piece of metric music (in pulses per second), we can then analyze the rhythmic events in that music in relation to the tempo. Just as we saw the frequency ratios in the pitch parameter form a useful means of comparison of one pitch to another, we may also use ratios in comparing rhythmic events. The comparison of one pulse to another is a starting point; if we take the tempo pulse as 1/1, then a pulse twice the speed of the tempo would be called 2/1. For example, a pulse indicated by a particular rhythm might contain three articulations in the space of two beats of the original tempo (3/2), or seven articulations in the space of five tempo beats (7/5) or virtually any other combination. In this manner, a wide variety of pulses can be indicated, although one is limited by his or her capacity to hear and produce distantly related pulses accurately.

It should be noted that a pulse is usually determined by the distance in time between the articulations of a passage of music, not the length of the durations. It is common that the successive sounds in a pulse may have varying duration, yet the pulse is felt as regular. For this reason, and to avoid confusion of the ratio relationships between durations (where 2/1 would indicate a duration twice as long as the first, and therefore half the speed in a pulse), the discussion of how one duration is related to another is left out of this essay. Snyder points out that there is a hierarchy of the different aspects of rhythm, and the duration in metric music is most often subservient to the pulse that the articulation of a sound indicates (2000). In this way, the function of duration in metric rhythm is similar to that of the envelope in timbre, since the articulation becomes more important than the sustained section of the sound.

In terms of rhythmic structure, a pulse might be called a *regular rhythm* if some type of sound articulates all of the individual beats. The term regular rhythm is also commonly used in musical terms to indicate a series of articulations that are related to the original pulse by a factor of two (2/1, 4/1, 8/1, or 1/2, 1/4, 1/8 etc.). This is analogous to the pitch phenomenon of an octave, where two pitches are deemed to have a special relationship if the interval between them is a factor of two. An *irregular rhythm*, then, occurs when not all of the beats in a given pulse are articulated, or when another articulated pulse is related to the original tempo by a factor other than two. Non-regular pulses are also called *polyrhythms* or *subdivisions* since they strongly imply a different tempo running parallel with the original. The term polyrhythm can also indicate another pulse implied by the *accenting*, or increased loudness in relation to the general loudness level, of non-regular groupings in a regular pulse. An unarticulated beat in a pulse is called a *rest*, and the building up of rhythmic patterns or phrases by combining beats and rests on different pulses forms the basis for many styles of musical composition (see **Section 8**).

Rhythmic Notation

An additional word here about the means of expressing durations and rhythms in traditional western notation is in order. In this system, the beats of a particular pulse in relation to the tempo are indicated by a set of visually differentiated noteheads and rest symbols. The noteheads, which are sometimes altered with auxiliary symbols indicating nuances in the envelope (such as > for accentuation), also give an indication as to the duration of each individual sound. Noteheads and rests may also be dotted to the right to indicate an increase in duration by half the original value, but they are named by the unaltered duration value (whole note or rest, eighth note, etc.), regardless of whether or not the sound played receives that duration when played. Also, the noteheads and rests are limited in that they only indicate the *regular* pulses of the original tempo. Any other pulse must be written as its closest regular pulse and enclosed within a bracket indicating the actual rhythmic ratio. Such a bracketed pulse grouping is known as a tuplet, or by a combination of the closest pulse name and the tuplet grouping (quarter-note triplet, half-note quintuplet, etc.). It should be clear from this description that this notation system is a very inexact way to notate any but the regular rhythms of a given tempo. However, a new system of rhythmic notation has yet to take off with the general public (Stone, 1980).

The complexity of visually representing fine gradations of pulse and rhythm is directly related to the numerous combinations of structuring durations of sounds and rests, especially when attempting to integrate this parameter with pitch or loudness or timbre in a coherent notational style. As difficult as this task may be, however, the benefits of utilizing various strata of time within a section or piece of music lie in the increased awareness of the overall sonic parameter of time. Without the element of time, none of the other parameters of sound can be articulated, and it is therefore safe to say that time is the most important element in music. This is reflected in the fact that time plays an important factor in all aspects of music, from the millisecond timing of fluctuations in a single sound’s envelope to the length in time of sections or whole pieces of music. This crucial role of time, then, is what integrates all of the sonic parameters into distinct units of event, phrase, section, and form.

5. Parametric Analysis

In order to consolidate the numerous concepts used to describe the sonic parameters throughout Part One, and to lead into a more musical discussion of the operations applicable to these parameters, in this section we will examine some of the issues involved in analyzing the parameters of sound in music. One question that is raised by all of the acoustical measurements that one can use to classify sounds is how do these various aspects combine to form a unified sound in itself? While each of the measurements in the parameters of pitch, loudness, timbre and time can be useful for in comparing the sonic quality of single sounds, in order to place these in their appropriate context one must consolidate them into representations of each of these major parameters individually. In so doing, it is possible to begin distinguishing the parametric changes across the larger time scale of phrases and sections in a musical work. The analysis of these parametric changes can lead to the recognition of the operations used to create the music itself, and these operations can then be applied to original compositions and improvisations, which is the subject of Part Two.

Relative Density

When investigating the progression of single sounds within a piece of music, one attribute that seems most important is the rate of change (or morphology) in the other sonic parameters during the sound. This musical parameter of *relative density* can be measured and formed into a continuous scale where denser sounds have a faster rate of change, and vice versa. The scale of relative density is shown below:

Table 5.1- scale of density relative to the number of events (changes in any sound parameter) per second (eps) that occur in a sound.

low density		mid		high density
1/12 eps	←→	1/1 eps	←→	12/1 eps

Since this is an absolute scale defined in terms of clock time, it is closely related to the rhythmic concept of tempo. The difference is that density measures then timing of events within a sound's duration while tempo indicates the timing between single sounds in metric music.

At one end of the density spectrum, then, are constan*t* sounds, where a sound has little or no perceptible change over its duration. It may also be said that if a sound has one or more parameters that alternate between two values (such as a steady vibrato or trill), and all of the remaining parameters are static, it may also be grouped as a constant sound. It may be useful to weight these steadily fluctuating parameters with more importance in order to indicate that they cause a relatively higher density. At the other end of the density spectrum would be sounds where one or more parameters changes continuously over its entire duration. These can be gradient sounds, found, for example, in glissandi (in the pitch parameter) or crescendi and decrescendi (loudness). High-density sounds may also be variable, as in the "wah-wah" of plunger-muted brass instruments in the timbre parameter. This constant/ gradient duality will be of importance in the discussion of musical composition in Parts Two and Three, but for now suffice it to say that often musical sounds lie somewhere in between these two poles.

Cogan has described a method of tallying rates of change in the timbre dimension, and this may be useful in fusing the parametric changes in the analysis of density (1984). This method essentially assigns positive, negative or neutral values to each measurement under consideration, and then condenses the values of these measurements into a single meaningful value that represents the amount of change within the entire sound. While Cogan's measurements of sonic parameters are defined vaguely in terms of descriptions of opposing attributes of sound (centered/ extreme, sparse/ rich, level/ oblique, etc.), it is possible to substitute the more precise measurements from Part One to the same effect. The measurements of each parameter can be averaged to determine the amount of change in the pitch, loudness and timbre of a sound, and these parameters can be condensed to indicate the density of the sound as a whole. The amounts of change from one sound to the next in a musical work can then be compared in order to determine the rate of change within a passage, section, or across the entire piece. While it is beyond the scope of this essay to produce a full-scale analysis of an existing work, it is hoped that further research into this method will bear fruitful results.

Visual representations of acoustics in music

In addition to the method described above, it is also possible to use the techniques of the acoustical measurement of single sounds to analyze parametric change in the context of longer musical sections (Malloch, 2000). Any measurement that can be graphed over time may provide insight into the progression of changes in that parameter. As an example, the same measurement of the fluctuation in total loudness of a sound seen in the discussion of amplitude envelopes above could be applied to the observation of the change in loudness over an entire piece. A less exact method of observing he overall variation of the loudness parameter can be found in the graphic waveform representation common to most audio editing software. Similarly, the graphing of sharpness, harmonicity or timbral width over the course of a musical work could aid in the recognition of the changes in the timbre parameter.

There is also unique visual tool developed by acousticians and speech researchers that offers the advantage of representing multiple parameters simultaneously. This is called a sonogram, which is the created by plotting the frequency spectrum on the vertical axis of a graph against the passage of time on the horizontal axis. Additionally, loudness is represented by the relative shading of the component frequencies of the sounds, where louder sounds are brighter and vice versa. The opening passage of music from one of the solo trombone pieces described in Part Three ("Theedie", **Score #4**) is shown in **Figure 5.1**. Notice how the harmonic spectra of the trombone tones stand out, as do the

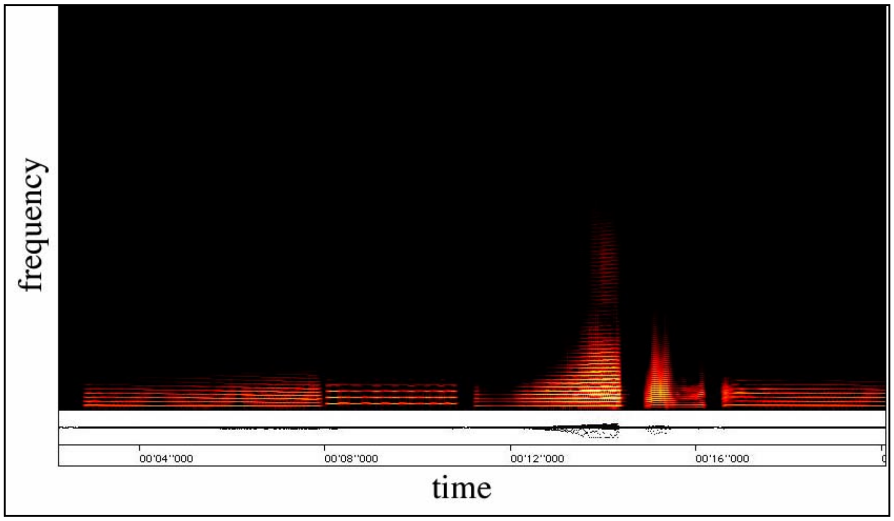

Figure 5.1- sonogram of the opening phrase of "Theedie" (Score #4).

timbre changes that occur with the increase in loudness of certain sounds. Many interesting features of the sonic parameters of musical sound can be readily identified, including formant location, brightness, loudness variation, relative timing of events, and more. Most sonograms, however, do not provide the means for measuring these attributes exactly, and thus they are useful only to a general discussion of broad-based changes in a piece of music. Acoustical measurements that depend on the computation of values derived from a sound’s spectrum, such as timbral pitch or sharpness, are not easily discernable on a sonogram either. In general, though, sonograms provide a useful way to quickly analyze a musical passage for its changes in the four major parameters of sound.

Part Two: Compositional Materials

6. Overview

Now that we have an understanding of how to describe and interpret some of the many parameters of sound heard in music and everyday life, we can look at how these parameters are manipulated to create a piece of music. In fact, by our new standards of acoustical analysis, music can be defined as the manipulation of the various parameters of sound in order to create a composition or improvisation. It can also be said that, in much music, these parametric manipulations are what generate the interesting features of a work (one exception would be certain forms of conceptual art pieces that are based on a single, unchanging sound, yet in the world of music these are somewhat rare). When one is developing a piece of music, one usually applies *operations*, or changes in value according to a specific plan, to one or more parameters until the material seems complete. Such operations might include varying pitches by a specific pattern, or creating a distinct rhythm from a series of different articulations or instruments, etc. It is important to note that the composer/ improviser may have these concerns in mind when designing his/ her piece, or may indeed have a vague notion of the musical processes taking place. In this sense, the definition of analysis precludes that of composition; since musical works include specific designs for varying the sonic parameters over time, then this same process must also be used to create a work.

In broad terms, there are two general processes used in the creation of music. First, there is the use of some theoretical rule or rules that defines what sounds are to be created at any particular time. For the purposes of this essay this may be called a *generative* musical process. If one were to limit oneself to creating sounds on a specific pulse (say, eighth notes) or with the specific pitches of a scale, this could be termed generative music. The other general process that is commonly used to create music is the application of a set of transformations to a specific (usually pre-determined) piece of sonic material. This may be called a *transformational* process, or *material transformation*. In many styles of Western music, it is common to develop a sequence of pitches (a melody) by playing them backwards, or upside down, or from a different starting pitch, and this would be an example of material transformation. Of course, a mixture of these two approaches can be utilized within a single piece of music, and often is. Material that is generated by a generative process can then be developed according to the techniques of material transformation, which in turn may suggest new types of generative processes.

7. Pitch Operations

As in Part One of this essay, we will begin our analysis of the compositional operations applied to the sonic parameters by examining the parameter of pitch first. Again, as noted in the analogous section of Part One, pitch has been the primary focus within the tradition of music theory. Also, much musical analysis simply (or not so simply, as the case may be) attempts to reduce the sonic information available in a piece to a discussion of the relative functions of the pitches involved. However, the means that musicians and theorists have developed to vary the sounds of a work along the pitch parameter can be quite sophisticated, especially in comparison to the less developed parameters of loudness or timbre. Those musicians that do bring complex designs to a parameter other than pitch often adapt the operations used to vary pitch for their own purposes. This section is meant to serve as an introduction to the kinds of operations available in pitch space, as a lead-in to discussion of other parametric operations, as well as to illustrate the difference between pitch and the other sound parameters.

Contour Space

We saw in Part One that tuning systems divide up the frequency spectrum into discrete steps, from low to high. According to Morris, this is all that we need to begin working with the most basic form of pitch recognition, contour space, or *c-space* (1987). In c-space, one works only with pitches and the contour shapes that moving from one to the next create, the distance between pitches (interval) is undefined. It is interesting to note that listeners who have little or no musical training (or musicians that are unfamiliar with a different tuning system) essentially hear pitch movement in this way, where each pitch is categorized as being either higher or lower than the preceding one. One can apply precise operations within c-space, however, and for this reason it could be useful to trained and untrained musicians who want to work with pitch.

The work of Morris, Rahn, Wuorinen and other music theorists in academic circles attempts to explain the functions of musical pitch by creating concrete analogies with the mathematical field of *set theory*, which analyses the properties of distinct sets, or groups of numbers (1987; 1980; 1979). Since groups of pitches can be easily represented by integers, as is the common practice in this style of music theory, general properties of these sets, such as interval content and possibilities of certain operations, can be explained by the model of set theory. A set is simply an unordered group of pitches, as distinct from a row, or segment, which is a group of pitches in a specific order.

According to set theory, the pitch operations available in c-space are inversion and retrogression (see **Table 7.1**). *Inversion* is the operation that essentially turns the original contour upside down, beginning on the same starting pitch. One useful benefit of using integers to represent pitches in theoretical terms is the fact that integers can be transformed by the use of mathematical formulas, thus in order to invert the contour of a pitch set in c-space, one can alter each pitch by the formula: $IP(n) = [q-1-P(n)]$. What this means is that in a c-space of q number of pitches, to find the inverted pitch one must first subtract one from q and then subtract the number of the pitch. Doing this creates a negative mirror image of the original contour. The *retrograde* form of a contour is simply the original sequence of pitches in reverse order, thus creating a backward contour. These two pitch operations can be combined to create another unique form of transformation, the retrograde-inversion.

Table 7.0.1- the effects of inversion and retrograde on a pitch contour in c-space.

pitch number							pitch number							
	7								7						
	6								6		X				
	5					X			5	X					
	4				X		X		4			X			
	3								3						
	2			X					2				X		X
	1	X							1					X	
	0		X						0						
		0	1	2	3	4	5			0	1	2	3	4	5
		order number								order number					
		original contour								**inversion**					
pitch number	7							pitch number	7						
	6								6					X	
	5		X						5						X
	4	X		X					4				X		
	3								3						
	2				X				2	X		X			
	1						X		1		X				
	0					X			0						
		0	1	2	3	4	5			0	1	2	3	4	5
		order number								order number					
		retrograde								**retrograde inversion**					

<u>Other Pitch Spaces</u>

In order to develop pitch material beyond the relatively limited field of c-space, the intervals between the pitches in a set are quantified, setting up a chance for two different possibilities. If the intervals between all of the adjacent pitches in a c-space set have equal ratios, and the pitches themselves are ordered from low to high, then *a p-space* is defined. As we saw in Part One, the Equal

Temperament tuning system currently in popular use can be defined as a p-space, and as such has distinct properties that can be developed for creating music in the pitch domain. One unique operation that can be applied in a p-space is that of *transposition*, where the original contour of a pitch set is mapped higher or lower in the p-space, retaining its contour while beginning on a different pitch. A further development is possible if we bring in the concept of equivalence; if each pitch in p-space is related to others by a modular interval (in the case of Equal Temperament, the octave), then all related pitches are said to be functionally equivalent, and a *pc-space* is defined. In a pc-space all of the pitches related by the modular interval are called a *pitch class*, and each pitch can be used in substitution for any other. Intervals other than the octave can be used as a modular interval in a pc-space, however this practice is not common. In practical musical terms, octave equivalence gives the composer/ improviser choice in choosing what register he or she wants the pitch to be sounded in, given the limits of the instrumentation at hand. The resulting choices will have a strong similarity with the same set of pitches sounded in different registers. Since all of the pitches in a pc-space repeat at the modular interval, a cyclical relationship is set up, as opposed to the distinctly linear p-space. Because of this, there are further complex relationships that may be examined and developed in pc-space than p-space. The operations of inversion, retrograde, and transposition (which apply to a pc-space), however, being the most commonly used across many different musical styles, form a set of basic tools for developing a group of pitches, and thus we will not examine any further set theory operations in this essay.

Of course, if the intervals in a c-space are measured and determined to have unequal ratios between adjacent pitches, a different set of properties is applicable, although the same operations are often possible. If the elements of a pitch set are ordered from low to high, an *m-space* is defined, which is much the same as a p-space, except for the unequal intervals. With octave equivalence, a *u-space* is set up, and this cyclical pitch space is analogous to a pc-space with unequal intervals. It follows that m- and u-spaces can be derived from a p- or pc-space by selecting a subset of pitches that have unequal intervals between them, but m- and u-spaces can also be formed by utilizing different tuning systems, such as Just Intonation or Pythagorean tuning. When the specific relationships of the pitches in the above m-, u-, p-, and pc-spaces form a pattern of intervals and none are greater than a major third, this is called a *scale* in common musical terminology. In Equal Temperament, the entire collection of pitches in a p- or pc-space is known as a *chromatic scale*, and another equal-interval scale, known as the *whole tone scale*, is formed by skipping every other pitch. Any scale that contains a repeating pattern of one or two intervals within an octave may be termed a *symmetrical* scale, and these can be defined by the intervals of the scale and the order they appear in. An *asymmetrical scale*, then is one defined by a m- or u-space, and the pattern of unequal intervals defines their shape, although usually they are named by some other means (by country or region of origin, after the person who "discovered" the scale, etc.).

Within a m- or u-space, the operations of inversion, retrograde and transposition are available just as in p- or pc-space, but the original contour or pitch set is mapped onto the unequal intervals of the pitch space, creating an inexact transformation of the original. Given the possibilities of combining the twelve pitches of Equal Temperament to form distinct scales, it should come as no surprise that there are literally thousands of scales, each with a unique pattern of intervals. Normally a scale is defined beginning on one pitch and including all pitches within an octave from that starting pitch. In order to use these pitches in musical contexts, however, musicians have developed other ways of imagining the same scale. By separating a scale into its component *modes*, each starting on a different pitch of a scale and extending to the octave of that starting pitch, new relationships are created out of the same grouping of pitches (Dallin, 1974). Scales that consist of only one interval, such as the chromatic or whole tone scales, can also be separated into distinct modes, however the distinction between them is vague to the ear when both the intervals and pitches of a set of modes remains invariant. Order is the important element in a mode, as the same pitches or pitch classes are present in the modes of a scale. For this reason, the modal way of thinking can also be used to effect with the (presumably non-scalar) pitches in a row or segment. Set theorists also call the process of deriving modes from a pitch segment *rotation*, since the order of pitches in a segment is offset by a constant value with each cycle in order to create the next mode (Krenek, 1960).

Tonality

One phenomenon that occurs within pitch spaces that is also of fundamental concern for many styles of music is that of *tonality*. Because of the multiple meanings this word has accumulated over time by many musicians it is difficult to determine a precise definition, but in general tonality is a function of an m- or u-space, where the unequal intervals set up expectations of pitch motion in the listener (Snyder, 2000). Any method of designing pitch structures where these expectations are deliberately avoided is thus known as *atonality*. It is common for musicians to speak of tones of a scale "resolving" to other adjacent pitches. This "tension and release" is an important factor in tonal music, and the effect is probably intensified by the usage of only a few scale types in a wide range of musical styles over a long historical period. The *major scale* is perhaps the most common scale used by musicians, and its characteristic pattern of M2-M2-m2-M2-M2-M2-m2 creates strong tonal resolution towards the first (or root) and fifth steps of the scale, as does the unaltered minor scale, with its interval pattern of M2-m2-M2-M2-m2-M2-M2 (Russell, 1958). Almost any type of scale, including symmetrical scales, can set up tonal references, especially when the tension generating pitches are inflected by changes in other sound parameters. Symmetrical scales can also be used to set up tonality by emphasizing certain pitches over others. By the same token, it is possible to use the pitches of a tonal scale in such a fashion that the tonal relationships are obscured (Snyder, 2000).

In order to create more tension (and consequently, greater resolution, or a generally heightened emotionality) in their musical works, Western composers beginning in the nineteenth century began to use non-scalar chromatic pitches within tonal contexts. This use of *chromaticism* eventually became so prevalent that the idea of tonal reference was dropped altogether in the beginning of the twentieth century, leading to the concept of atonality. This progression may seem counterintuitive given the above description of tonal scales (m- or u- space) being derived from the chromatic scale (p- or pc-space). By this model, tonality is a more advanced form of atonality, since the pitches of one are contained within another. As we saw, however, m- and u-space scales can exist without a larger p- or pc-space, and therefore tonality is something distinct which can be found either on its own or within an atonal framework. Also, the concept of equal temperament was developed in order to accommodate the modulation (or transposition) of tonal music to every pitch in a tuning system (Partch, 1949).

To sum up the above discussion of the musical uses of pitch, the defining of a pitch set or c-space contour is an example of a generative process, and this may be done by a number of means, whether intuitively, by chance, or in order to achieve a specific outcome. When one defines a pitch set in pc-space, it is also possible to determine the interval content of that set, or what types of intervals are available between the specific pitches, and how many of these occur. When writing music, composers often choose a set based on the interval content and other properties that fulfill the desired goals of the composition as a whole. The set is then developed according to the above and possibly other operations, and tied in with other parameters, such as dynamics and rhythm. The key concept here is that the musical material created by a transformational process is uniquely related to the original material, thus creating cohesiveness within music based on such transformations. There are many types of operations that can be applied to pitch and other sound parameters, and a composer/ improviser may find a personal approach by developing or defining unique operations within their own music (Wuorinen, 1979). A composer may also find distinct means of creating generative musical processes using the various pitch spaces defined in this section.

Intervallic Operations

Thus far we have examined the use of distinct groups of pitches or pitch classes, and observed the function of transformational operations on the pitches and intervals of a set, yet intervals can also be utilized as compositional elements themselves, without regard for the specific pitches that intervallic motion creates. This is a sort of negative c-space (i-space?), where the interval content of a set is defined, but not the pitches. Working in this way with intervals, one can utilize generative, as well as transformational, processes. A specific pattern of intervals may be mined for its unique content and used to generate an ever-changing contour by repeating the order or content of the interval set with free choice as to whether the interval will move to a pitch above or below the preceding pitch. As an example, a distinctive sound can be found by beginning on any pitch and moving to subsequent pitches by increasingly larger intervals, and this pattern may be repeated or reversed when the limits of an instrumentalist's range are met. Similarly distinct sounds can be generated by other interval patterns that have unique characteristics.

In addition to the above and other generative interval processes, there are also transformational operations unique to intervallic manipulation that are not found in pitch space. Because of the cyclical nature of pc-space, it has been established in musical set theory that not only is the octave a modulating interval in pitch space, but the diminished fifth/ augmented fourth interval creates functional equivalence among pitches as well. An example will help to illustrate this theory; whether one moves up a minor seventh or down a major second from the pitch C, a B flat is reached. Since the relationship of a pitch x to another pitch y may be defined as moving up or down from either pitch to the remaining one, it stands that the interval between these pitches can be determined by either x-y or y-x. Thus major seconds and minor sevenths are equivalent, as are the complementary intervals within an octave up to the augmented fourth (Rahn, 1980). Because of this fact, a unique transformational operation is available to apply to an interval pattern, that of interval *inversion*. In this process, all intervals are converted to their complements, again creating a new set distinctly related to the original, with a completely different sound.

Another important set of transformational operations can be found by *expansion* or *reduction* of the intervals within a set by a constant amount. This is distinct from the multiplicative operations defined in Rahn's and Wuorinen's texts, where pitch space is defined by ascending integers around the circle of fifths, rather than ascending the chromatic scale, and the pitches of the set undergoing multiplication are mapped onto this new space, (1980; 1979). Such an advanced type of operation, where essentially only certain intervals are inverted while other others remain the same, is clearly less "hearable" than operations that utilize a 1:1 relation of pitches, yet this process is conceptually valid and thus may be useful as a compositional tool. The degree of audibility achieved by a given process or operation is an effect that can certainly be determined by the composer/ improviser's sense of aesthetics and training. It is also an important issue to develop an understanding of what operations produce what types of reactions in a group of listeners. The highly subjective sense of what types of sound create what emotional or psychological effect in a listener has yet to be explained in a scientific manner, and in fact this may not be possible given the wide range of reactions that each individual may have to a given sound. This issue thus lies outside of the scope of this essay, but the rigorous classification of all of the parameters of musical creation is central to Sonogrammatic theory, so treatment of this subject may be developed in further research.

8. Rhythmic operations

We saw in Part One that a major distinction within the sonic parameter of duration occurs between event-based time and metric time. This distinction also governs what types of operations are available for manipulation within these two types of rhythmic structure. For the most part, any operations applied to music in event-based time are generally applicable to any parameter where the specific order of a sequence of events is transformed, and so we will go more into depth about these operations in the discussion of compositional form below. However, it should be noted that if the sounds that occur in a passage of event-based music are measured according to seconds or any other time scale, then that passage can be conceived of as having a metrical structure, regardless of the intent of its creator. This is roughly analogous to what we saw in pitch space, where a less precise definition of c-space only involves deciding what the sound events (pitches, in that case) are, then this model becomes more complete by determining the intervals between the events. We saw also in Part One that any metrical tempo can be scaled relative to the absolute framework of clock time, so this conception of event-based time is consistent with the ideas of density and rates of speed developed previously. We will call this special case of the measured timing of sound events *scaled time*. In any case, it will become clear in this and the following sections that many of the operations defined in relation to the pitch parameter are applicable to rhythm and other sonic parameters, but it is also beneficial to find the unique operations available only to a specific parameter.

Processes Related to Pitch Functions

One set of rhythmic operations that is applicable to both metric and scaled time is also analogous to operations defined in our discussion of pitch in **Section 7**. Expansion and reduction can occur within the rhythmic domain by lengthening or shortening the overall time of a particular piece of musical material, while keeping the relationships between the sound events and their durations proportional. This is easily done to recorded music with most audio-editing software, yet for musicians it is all but impossible to stretch the temporal envelope, so only the option possible is to alter the timing of events. In metrical music, expasnsion and reduction are known collectively as *metric modulation* if the underlying tempo of a musical passage, which as defined earlier form the pulse ratio 1/1, are converted to a different pulse ratio. By this means the pulse modulated to becomes the new 1/1 tempo, and the corresponding duration of each sound event is multiplied by the converting pulse. By this process, for example, when expanding from a 1/1 to the 3/1 pulse, a half note becomes one partial of a quarter note triplet (1/2 X 3/1 = 3/2), or eighth notes become sextuplets (2/1 X 3/1 = 6/1). Expansion is defined when the modulating pulse is greater than the original, and reduction when the modulating pulse is lesser. These processes can be used in practice as either transformational, by multiplying the spacing of attack points in a melodic or rhythmic sequence by the corresponding ratio, or generative, by altering the relative tempo of music improvised in other parameters.

In fact, rhythmic expansion and reduction can be achieved through very simple means; by changing the overall tempo of a passage, the attack points and durations of the sound events will be multiplied by the appropriate ratio. A contrasting rhythmic process would be to keep the tempo steady and change the meter of the music. However, due to the inherent patterns of accentuation found within specific meters, the "feel" of this method is audibly very different from the application of a modulating pulse, and this would not be termed metric modulation. Meter change where the basic tempo remains constant changes the relationships of strong and weak beats implied by the meter. The manipulation of these unequal emphasis patterns is what generates tension and release in the rhythmic domain, just as we saw the unequal intervals of tonality do to generate tension and release in the pitch domain. Another transformational rhythmic operation that affects a change in the accent patterns of a piece of material, even in the same meter, is related to a pitch operation that we saw in the preceding section. Rotation of a piece of material rhythmically occurs when the material is offset or displaced by a constant value, shifting sounds from strong to weak beats, and vice versa.

The other pitch-based operations that we looked at in **Section 7** do not necessarily apply to rhythmic space, and for a specific reason. In order to invert or transpose a piece of musical material in the pitch parameter, a sound event within a set must have a value as well as a place in relation to the other elements of the set (interval). In rhythmic space, there are no meaningful values (as we saw in Part One, duration plays a minor role in our perception of rhythm), only the relationship of time intervals between the attack points. For this reason, attempting to invert or transpose a sequence of attack points and durations is not meaningful, and, as we suggested in the preceding paragraph, the retrograde form of a rhythm implies reversing the temporal envelopes of each sound. It is possible however to reverse the order of attack points and relative durations in a sequence. Perhaps the concept of equivalence can be used to further explore the analogies between inversion and transposition of pitch and rhythm. For now we will turn to examining the operations that are unique to rhythmic structure.

Unique Rhythmic Capabilities

While repetition is a process that can be applied generally to any sequence of events, its application to the rhythmic parameter of music creates a few distinct possibilities. Some of these will be explored in more detail in the discussion of form below, but for now suffice it to say that the repetition of musical material can be either strict or loose. A *vamp* is a common vehicle used in improvisatory music, and is accomplished by repeating certain elements of the musical material (usually the bass line) while other variations are developed either simultaneously (in an ensemble) or within the rest points of the original material. Many musical processes in other parameters can be made more apparent by repetition of the material utilizing the desired process, and in fact repetition is one of the unifying aspects of our perception of the important elements of a musical passage (Snyder, 2000; Tenney, 1988). One of the perceptual aspects of repetition is that it creates an expectation of an endlessly repeating cycle, which is often departed from in order to affect tension and release. Thus, repeating certain elements of a musical process shifts attention away from these elements, so that other aspects of the music can be focused on (Snyder, 2000).

Thus far the operations that we have examined in this essay have been culled from many sources and are commonly used in well-known musical styles, but due to the fact that there are few distinctly rhythmic operations, I will introduce one that I have personally developed. We saw in **Section 2** that a pulse that is a subdivision of the main tempo (and that tempo pulse itself) can be implied without necessarily sounding each pulse. In this way it is possible to conceive of every pulse implied by a passage of music as being available for sounding or resting. By this process of *binary variation* one can transform a piece of musical material, substituting sounds for rests and vice versa. The total number of possible combinations of a passages tones and rests is a function of n! (read "n factorial," where 1 X 2 X 3 X… n), where n is the total number of implied notes. This process of treating sounds and silences as equivalent is perhaps similar to other equivalence operations in other parameters, but there is no direct corollary with other parametric operations.

It is often a tactic that composers who want to focus on certain parameters in a passage of music will de-emphasize other parameters, and we saw that rhythmic repetition can be used for this purpose. However, it is also possible to de-emphasize a given parameter by making all values in it the same, and in the rhythmic parameter converting all of the sounds in a piece of musical material to a pulse does just this. In this way the values in other parameters of elements in a sequence are heard more in their own right, especially if more than one parameter is neutralized. By this process, the changes in pitch or dynamic level of a musical passage may be isolated from an original passage that was highly rhythmic. Of course the reverse is also true that music that emphasizes rhythm can be developed from just a string of pulses (as defined in a pitch set, for example). The many compositional operations that we have examined thus far have been either applicable to multiple parameters, or found only in a single parameter. The above process of emphasis and neutralization actually clarifies the distinction between the parameters of sound, and provides a means to musically explore each parameter singly or in combination with other parametric variations.

9. Secondary Parameters

The preceding sections of Part Two should show that there are a healthy variety of operations to choose from when varying music along the pitch and rhythm parameters, many of which are in common use among musicians today. Unfortunately, there are not as many operations that have been developed for the variation of timbre, loudness and other "secondary" parameters, like density or physical position or harmony, etc. (Snyder, 2000). Musicians have been developing techniques of variation in those parameters, however, for quite some time, mostly by intuitive means. This is not to say that musicians have not developed rigorous systems for variation of secondary parameters, since they certainly have in the field if electronic music, just that there isn't a standard set of techniques that are commonly used to vary timbre and loudness that crosses over musical styles. However, there are ways to apply pitch and rhythm operations to the secondary parameters that we will look at below. Just as we saw that the parameters of timbre and loudness have been the least understood in the field of acoustical analysis, it seems that these parameters have also been the least developed musically. Given all of the attention shown to these secondary parameters in theoretical circles since the last half of the 20th Century, no doubt more operations will become better known as time progresses.

Extended C-Space Operations

One interesting aspect of the scales of measuring timbre and loudness in Part One was that each is designed to classify sounds with a particular value within its own scale. The spectral measurements of sharpness and roughness, or the loudness measurement of phons, all plot the values of sounds on a single axis from some low number to one higher. This is reminiscent of the process of defining pitches in c-space, and indeed the techniques of sonic transformation used on pitches in c-space can be applied to any single axis measurement. As we saw earlier, this gives one the capacity for applying inversion and retrograde to the contour of values that is created in c-space. **Table 9.1** shows the effects of inversion on a contour built from successive dynamic level variations. Of course, almost any values may be plotted within c-space and then permuted by transformational operations, but the techniques are the most audible when the progressive values in a scale increase or decrease in a gradient manner, rather than jump from one radically different sound to the next (Morris, 1987).

Table 9.1- inversion of a dynamic level contour in c-space.

dynamic level	ff					X		dynamic level	ff		X				
	f				X		X		f	X					
	mf								mf			X			
	mp			X					mp						
	p	X							p				X		X
	pp		X						pp					X	
		0	1	2	3	4	5			0	1	2	3	4	5
		order number								order number					
		original contour						**inversion**							

In practical musical terms, then, in order to utilize the c-space operations on a secondary parameter one must first be able to plot a scale of gradations and match this to the selected sounds of an acoustic instrument. Since we saw that the timbre of an acoustic instrument does indeed change with pitch or loudness changes, it is difficult to emphasize or neutralize many spectral measurements from a passage of music in order to develop a pattern only within that parameter. However, it is possible to plot some of the actions that musicians use to vary timbre along a gradient scale. An orchestral string player might, for instance, divide the distance between the fingerboard and the bridge into distinct zones of bowing, and progressing through these in turn would affect a gradient timbre change. Any linear parameter involved with musical creation may be plotted and transformed in c-space, which may be useful for those musicians interested in exploring spatial effects (like directionality, distance or reverberation) or complex electronic processing.

One offshoot of the development of atonality in Western music was a method for varying timbre in a linear fashion with an ensemble of acoustic instruments. This process, known as *klangfarbenmelodie*, or sometimes *pointillism*, involves hocketing the sounds of a piece of musical material from one instrumentalist to another, thus affecting timbre change independently of the loudness or pitch parameters. Varying the timbres available on a single instrument as shown above is analogous to this process. While klangfarbenmelodie has its problems as a model of organizing sound, especially since we saw that the relationship of timbre to other parameters cannot always be isolated. It may be possible to now use the timbre measures from Part One to create more exact forms of klangfarbenmelodie. These same principles might also be applied to the overall *orchestration*, or distribution of parts of the music to members of an ensemble, in order to plot a long-range timbre development. Such a method would require a large database for sound analysis where the significant attributes of a sound, such sharpness, roughness, loudness, etc., can be compared with a wide range of other sounds from the instruments that one is to work with. Many researchers have been working on ways to create such comparative sound analysis methods, so this goal may not be too far off (Foote, 1999).

Sound Color Operations

One theory of operations on a secondary parameter that moves beyond the single-axis representation of timbre measurement is the sound color theory of Slawson (1985). This theory, as we saw in Part One, attempts to define a subset of timbre, known as sound color, that is a function of the frequency of the formant resonances in a spectral envelope, thereby equating spectral envelope shape with the characteristic sound of vowels. While the application of sound color theory is only meaningful when working with weakly coupled instruments such as voice or electronic filters, the sound color space is of interest in that it is multidimensional. The frequency position of the first and second formants form the axes of this space, and the different vowel types are found within distinct regions of the space. Slawson also defines a number of dimensions within this sound color space based on his experience with speech theorists: *openness* increases proportionally with the frequency of the first formant, and *acuteness* increases proportionally to the second formant. By moving diagonally within the sound color space, it is possible to affect changes in *smallness*, and by moving from the outer regions towards the center, or vice versa, changes in the dimension of *laxness* are achieved.

Such a multidimensional space is a useful way of modeling the complex factors involved in our perception of timbre, and the vowel sound analogy is clear to those not familiar with musical issues. Showing that the operations of transposition and inversion can be applied to transform a pattern of sound color changes, Slawson further defines how these operations can be applied to the dimensions in sound color space. When these operations are applied to a sound color contour, the resulting new shapes are shifted along one axis or both (transposed), rotated, collapsed or expanded. By this means sound color theory treats timbre in the same complex fashion that is found in the pitch parameter and thus more operations are available to transform musical material in the timbre parameter. We saw in Part One that sound color theory is not useful for the strongly coupled systems of acoustic instruments, but it is perhaps possible to use combinations of other timbre measurements to create the same type of multi-dimensional space for complex transformational operations. It may also be possible to re-conceive other normally single scale parameters, such as loudness or density, in the same manner. Clearly much work has yet to be done in the fields of timbre research and manipulation.

10. Form

With the information contained in this essay up to now, we have a fairly rigorous means of defining the various sound parameters and their properties, and we are also able to apply operations, whether generative or transformational, to those parameters in order to create musical material. To develop a unified composition or improvisation, however, the musical material must be worked into a *form*, or arrangement of events that govern the flow of music from one sound to the next. The possible types of forms are at least as numerous as the many parametric operations that we have discussed, and also like these the creation of unique forms is a trademark of the serious composer. With this section, the process of creating a piece of music from the ground up is complete; the product of a parametric operation is a phrase of musical material, and these can be varied and combined with other phrases to form a section. These sections can subsequently be varied and combined with others within the overall form of the entire piece. Form is one aspect of musical structure where it is normal and even necessary to speak of general principles behind the music in an exact manner, unlike how some of the basic parameters of sound itself have traditionally only been vaguely defined. Many parametric operations are based on the procedures of working within a particular musical style, and it is thus also hard to speak generally of these. Form brings these elements together and yet is also it own entity, and as such acts like an empty vessel into which the liquid elements of sound are poured, supporting its contents while having unique features of its own.

Sequences of Events

Before we look at form from the higher-level component of the section onward, we must first finish our discussion of the operations that create *phrases*, or unified collections of single sounds, by examining the processes involved in the transformation of a group of sounds. Earlier in Part Two these operations were defined within each of the major sound parameters, but it is also possible to create sequences of sound events that emphasize different parameters from one sound to the next. It is useful to determine the properties of these sequences, as well as the transformational operations that can be used to vary the order of sequential events in a way that is related to the original sequence. To understand fully the properties of sequences, it is beneficial to first examine what can happen between two sounds that occur in order.

For any given parameter of a sound event, there are three distinct relationships that it can have with the sounds immediately after it. If all of the values of a sound remain the same in the next event, then a strict repetition has occurred. A parameter may also change values, and the new values may be either constant or gradient as well. The more parameters that remain constant, the more a sound seems to be a repetition of its predecessor, and **Table 10.1** illustrates how parameter change can vary along a scale from strict repetition to complete change. Duration and rhythm are also major factors in the repetitiveness of sound events, for equally spaced intervals between attack points imply a repetitive pulse in the rhythmic parameter. By the same token, the equal duration of sound events in a sequence implies repetition even with changing time intervals between them, as long as the interval isn't greater than our capacity to remember the preceding sound (Snyder, 2000). Once the properties of the relationships between sound events are known, it becomes possible to create sequences of sound events that manipulate the constant/ gradient polarity, and vary along more than one parameter at a time.

Table 10.1- the distinct relationships that can occur between temporally adjacent sounds, on a scale from strict repetition to complete change.

degree of change	**gradient parameter(s)**	**constant parameter(s)**
strict repetition	same	same
↑ / ↓	new value(s)	same
↑ / ↓	same	new value(s)
↑ / ↓	become constant	same
↑ / ↓	same	become gradient
↑ / ↓	become constant	new value(s)
↑ / ↓	new value(s)	become gradient
complete change	new value(s)	new value(s)

The definitive aspect of a sequence is the ordering of its component elements, and this ordering can be conceived of as a parameter for manipulation similar to the others described earlier in Part Two. Since the place in order of each element can be represented by a number from low to high, it follows that this single scale parameter can be transformed by the c-space operations of inversion and retrograde, and possibly other operations not covered here. One must take care that the effects of operations on other parameters aren't negated when performing sequential operations, and this statement is true of other combinations of operations as well. For the purpose of varying the parameter of order found in sequences, it is assumed that the sounds within that sequence have different values in two or more parameters, otherwise operations on that sequence may be confused with operations in a different single parameter. We saw in Part Two that normally if one wishes to isolate a certain parameter they must leave the other parameters constant, but the case of sequential ordering appears to be the opposite.

One other basic set of issues is fundamental to the way we perceive form and these refer to how the changes in sound parameters indicate the grouping of sound events into distinct musical phrases. Both Tenney and Snyder draw on the field of Gestalt psychology to answer the question of what indicates the closure of a phrase, to which there are a number of responses (1988; 2000). The three major factors of grouping are proximity, similarity, and continuity. The principle of *proximity* in this procedure applies to the parameter of time; sounds are grouped together if the distance between them in time is relatively close. With *similarity*, given equal time intervals sounds are grouped according to changes in other parameters. A stream of notes that suddenly becomes much louder (or brighter, higher-pitched, etc) forms a new grouping with this change in loudness (or other parameter), for example. If a sequence of sound events forms a clear pattern of moving in one direction within a parameter, then a break in that *continuity* by leap or reversal of direction also causes groupings to form around either side of the break point. There are other factors that play into our segregation of sounds into groups, such as cues from earlier material within a piece, but for our purposes it is enough to note that parametric change often affects our perception of how sequences of sound events are formed into phrases. These grouping principles can of course be used to good effect whether the musician's intent is to emphasize or obscure the perception of phrases within music.

Block Forms

Now that we have an understanding of the formal processes that are involved in the generation of phrases, we are able to delve further into form and examine the means by which sectional music is created. Of course, one of the simplest musical forms is single-process music, where an operation is defined in one or more parameters, and carrying out this process completes the entire piece. Generative processes, especially those involving some sort of improvisation, are perhaps the most effective examples of single process music, unless there is a lengthy enough set of material that one may apply a transformational operation to create a complete piece. In any case, single process music represents the most basic type of *block form*, which is the umbrella term given to any form that consists of distinct sections. What forms sections however is a change in the overall processes of music, whether this consists of developing groups of similar phrases or switching between generative processes, etc. It is possible to create clear or vague sectional groupings in much the same way as described above for forming phrases, and again this choice is a matter of the musician's personal taste.

The rules for creating variation of sections within a form is much the same as we saw for developing sequences or phrases. If a piece of music is determined to have just two sections these can either alternate or repeat, but forms with three or more sections may be varied using the c-space operations of inversion and retrograde. However, unlike with sequences, where events must be varied along two or more parameters to avoid confusion with single-parameter operations, adjacent sections may be similar and the form of the musical work as a whole remains clear. Repetition is a strongly cohesive factor at the sectional level, for it increases the listener's ability to remember the musical content of a form. As the progression of elements in a musical work get increasingly larger, from sound event to phrase to section to form, the listener relies more on long-term memory to process the relevant features in the music (Snyder, 2000). Form, then, is one parameter of music that the creative musician might manipulate in order to affect the psychological state of the listener in a direct way, by playing with their sense of time and memory.

Gradient Forms

In contrast to the idea of creating musical pieces with distinct sections, it is also possible to work with complex forms that move continuously from one process to the next, and these are called *gradient forms*. It should be noted that, unlike our definition of gradient single sounds that move smoothly from one distinct value to another, the music in a gradient form might cover a wide range of values as long as the motion between them is continuous. One set of techniques that a musician can accomplish this with are *additive and subtractive processes*. These terms are taken from the field of computer music, where composers either build up a sound partial by partial or successively filter regions of a complex tone in order to create a specific sound. In terms of form, an additive process might consist of performing successive operations to one parameter at a time while retaining the previous operations. A subtractive process would reverse this procedure. These types of processes can be difficult for the performer to achieve on their instrument, but this is probably due to the relative lack of familiarity instrumentalists have with negotiating the secondary parameters. In ensemble music, however, additive and subtractive processes are much more familiar and even common, for the different operations (even multiple operations on the same parameter) can be distributed among the group of instruments.

Rule-based Structures

The final set of compositional techniques that we will examine for the purposes of this essay are rule-based processes, which are fundamentally different from all of the other types of form that we have looked at thus far. The reason for this is that the element of *improvisation*, or free choice, is factored into the progression of sounds in a piece of music. This principle of letting the performer decide some or all of the changes in parameters of sound events means that the composer has relinquished control over the parameters involved to assure that a degree of spontaneity is included in a section or musical form. The acoustical effects of improvisation on the parameters of sound cannot be calculated ahead of time, but it can be analyzed afterward, and doing this can be helpful in determining the desirable aspects of improvisation in each parameter.

Of course, the most extreme case of a rule-based process would be total or "free" improvisation, where all of the parameters of sound, including form, are left up to the musician(s). There are a wealth of other means to incorporate free choice into a musical piece, and ratio of composed/ improvised processes indicate the predictability of the music. When certain parametric operations are pre-arranged and others improvised, this is known as an *open space*. It is possible to play a piece of rhythmic musical material with "open rhythm", or "open timbre", and these directions imply that the parameter indicated should be manipulated by intuitive means. An open space can also be imposed on completely *deterministic* material, where all elements are composed beforehand, by isolating one or more parameters and replacing their values with improvised values.

There has been much debate over the validity of both improvisational and deterministic music, with musicians on the extremes of each position vying for the supremacy of their approach. This is an ideological debate that falls outside of the scope of this essay, yet it is assumed in these pages that the two approaches are equally valid, and can be combined furthermore. One special type of rule-based structure that lies within the domain of neither improvisation nor composition is that of the aleatoric process. *Aleatoric* music is that which is determined by chance procedures, such as a toss of the dice or flip of a coin. Composers have developed many unique methods for introducing randomness into their music, and many of their scores indicate the timing of events relative to a clock. This is an example of an event-based time that has been scaled, and as such the material provided in such a passage can be transformed by the operations indicated earlier. In essence, aleatoric music is highly deterministic, yet the music often sounds as if improvised due to the irregularity of the material. The reverse is also true; music that is completely deterministic can generate such a sensation of unpredictability that it sounds improvised or aleatoric. Like improvisation, the effects of aleatoric procedures on the parametric values in a passage of music cannot be predicted, but unlike improvisation, once the procedures are notated they are as predictable as any deterministic music.

Combining compositional approaches

The techniques of sound manipulation and combination outlined in Part Two are illustrative of the fact that the possibilities for the creative musician to create unique music are seemingly endless. A wide variety of parameters and measurements within those parameters are available for manipulation, from the smallest element of a single sound to the processes that govern the progression of an entire piece of music. Interesting pieces can be created with very simple operations on only a few parameters, or complex forms of simultaneous processes can be designed that are highly systemic. The process of utilizing the same types of operations in more than one parameter simultaneously is known as *integral serialism*. This method of highly deterministic music was first developed by 20th-Century composers who began applying the pitch operations of atonality and set theory to parameters other than pitch.

It was noted earlier that, if clarity of the musical processes underlying a piece is desired, care must be taken not to obscure the effects of one operation by other simultaneous operations. Varying one or two parameters at a time while keeping the others constant over time is conceivably the best way to ensure such clarity. For the improviser, this limitation is indeed necessary, for the effects of operations on multiple parameters often exceeds the cognitive processes of a musician working in this manner.

The choice of what types of operations, processes and forms to use in a given piece certainly depend on the composer/ improviser's personal sense of aesthetics, which may or may not be governed by allegiance to specific musical ideas or styles. Sonogrammatic theory attempts to look beyond these allegiances in order to develop a clear understanding of the general processes involved in musical creation. The concepts laid out in this essay, however, were chosen and developed from the author's own personal experience and taste, and for this reason are certainly biased. Many of the ideas herein are culled from respected sources in different fields of musical research, but there are many more sources not included that may help elucidate further relevant topics. Part Three will attempt to outline the development of Sonogrammatic theory in order to provide the reader with an idea of how the ideas within this paper can be used practically. It is my opinion that Sonogrammatics can be viewed as an approach to musical creation that can filter the disparate processes found across musical traditions and history into their component parts by analysis, in order to utilize these techniques in the creation of new and unique musical forms.

<u>Part Three: Practical Applications</u>

Many different activities and research approaches have gone into the development of the theory of Sonogrammatics. In this last part of the essay we will describe these as a means for pointing out the issues that arose during the period of development as well as the benefits and drawbacks of putting the theory into practice. While we will see concrete examples of how the concepts of Parts One and Two can be implemented in actual musical work, these are just some of the myriad possibilities that can be derived from Sonogrammatics. With more time the ideas proposed in this essay can be further refined and expanded, by incorporating new resources and by the familiarity that comes with more experience.

11. Instrumental technique/ sound analysis

The first stages in the development of the theory of Sonogrammatics came with the critical thinking and exploratory attitude of developing technical facility on an acoustic instrument. I have been studying the slide trombone for a number of years and working within many improvisatory contexts. At some point in my growth as an instrumentalist I determined that the conventions used in general music making were vague and inexact and I sought ways to remedy this situation. The first step was to catalog all of the various types of sounds available on the trombone and attempt to quantify these in some way. This involved practical experience as well as a review of literature on contemporary instrumental technique. Since the mid-nineteenth century (and most likely before), instrumentalists have been engaged in a constant expansion of the ranges of each sound parameter available on acoustic and electric instruments. Moreover, many of the techniques that they have developed to do this have become increasingly common (Stone, 1980). One of the ways that musicians improve these techniques is by the repetitive practicing of exercises designed to isolate certain musical elements, and the information in Parts One and Two hopefully suggests means of developing different types of exercises to explore all of the various parameters of sound.

One psychological aspect of the musical process of repetition, which we saw in the discussion of it in Part Two, is that the elements that are repeated in a sound are moved to the back of the listeners focus and other parameters are studied more closely. For this reason, as I worked with the conventional and extended techniques to develop facility in negotiating the sonic parameters I began to think more about their function in musical contexts. The similarities of these techniques across musical instrument families indicates that the concept of the independence of the sonic parameters is a general musical issue, although some music based on certain instrumentation neglects to utilize changes in every parameter due to personal taste or instrumental capabilities. The latter of these two reasons became the impetus to study just how far the various types of instruments can extend their sounds in each parameter.

To achieve this aim, I drew on my colleagues and associates in the instrumental music world, recording any constant or gradient sound types in each parameter, and combinations thereof, that were available. A representative instrument from each of the major families was sought out, as well as players of these instruments that had experience with extended techniques. Analyses of some of these sounds are incorporated into the structure of Parts One and Two of this essay, yet the details of all of the sounds recorded has yet to be mined fully. What was found is that each instrument (excepting the drum set) was apparently designed to negotiate the pitch parameter primarily with moderate values in the secondary parameters (the so-called "normal" mode of playing) and extreme values in most parameters are more difficult to perform. Also, the more extended techniques that are applied to a sound logically increases the difficulty of achieving the desired result, and some combinations of sound types that might be sensible in theory are impossible to execute. However, there are many types of sound creation via the different parameters that are relatively easy to execute which for some reason or other aren't commonly used by instrumentalists or called for by composers. By reviewing some of the literature about instrumental technique it is apparent that new approaches to parametric manipulation develop out of the necessity of achieving new musical concepts, and with time it is possible that the previously mentioned "nether-regions" of instrumentalism may be mapped more precisely.

Acoustics and the Computer

Concurrent with the study of instrumental negotiation of the parameters of sound, I began to search for definitions of the parameters themselves. After experimenting with different methods of verbal and symbolic description of sounds, common in the Western and other traditions of music, I turned to the fields of acoustics and auditory perception to take advantage of the relatively precise definitions of sounds that these fields provide. I began to work with the assumption that if the vague descriptive terms used to describe parametric values in music (such as dynamic level markings) could be replaced by terminology from auditory science, and this procedure would contribute to greater understanding of music, both generally and related to specific techniques.

This led to an interest in sound analysis, which is most effectively done with the digital techniques available on a computer, since mechanical and analog electronic methods of sound analysis are often highly variable. It is because of developments in computer science that greater precision of acoustical measurement can take place, for a sound sample that has been digitized is easily quantified numerically and the distinctive features of a sound can then be extracted by complex computational models (Roads, 1996). Many of the ideas about timbre expounded in Part One were formed by comparing the benefits of a number of measurements, including tristimulus methods, vowel formants, amplitude envelopes, etc. While some small-scale analyses of sound parameters was done for this essay with the recorded instrumental samples, what is needed to fully compare the instruments' capabilities is a searchable database of the thousands of sound types, so that specific measurements may be isolated and examined and more closely. Since there is no such existing sound analysis software that fulfills this need accessible from commercial or academic sources, attempts were made to develop such a tool. It should be noted that most affordable sound analysis software, including the ones used in writing this essay (PsySound and Amadeus), are limited by only being able to evaluate one sound at a time. Due to lack of programming experience and advanced scientific knowledge required to put together a useful database flexible enough to investigate the many issues of musical sound, this project is still in the works.

One useful approach found in acoustical experiments is to observe the changes in one parameter that occur with intended change in another (Levitin, 1999). Because of previous experiments such as these it has been revealed that the spectrum of a sound varies with changes in loudness or pitch, as we saw earlier. The use of a database of acoustic sound measurements would be useful to the further exploration of such links between any given parameter and others. It would also be useful as a compositional tool; to find ways to vary certain isolated parameters while leaving the rest constant is an effective musical strategy, as we saw in our discussion of musical operations. The ability to plot continuous and discrete variations in all of the sonic parameters according to the acoustical (as opposed to musical) definitions and measurements of those parameters is a long-range goal in the development of Sonogrammatics. Due to the complexity of the possible combinations of parameters, however, this may be a life-long endeavor, and so the current essay marks just a beginning in the evolution of Sonogrammatic theory.

12. Ear Training

The above description of the use of the current theory in sound analysis should indicate that Sonogrammatics is more than just a tool for creating radical forms of musical compositions. It is also an approach to thinking about sound and music in a systemic and logical manner. Another benefit of this approach is that it provides a unique means of *hearing* such musical sounds as well. Since music is first and foremost an auditory phenomenon, any complete theory of its working should include ideas related to our perception of sounds. To this end, Sonogrammatics provides a set of ear training tools that musicians may find intriguing.

If one is to perform complex and precise parametric changes on an instrument, it follows that the scales of measurement that have been defined for each parameter can be utilized in their execution as well as analysis. It is common for musicians to practice musical exercises and passages with tuners and metronomes to aid in developing greater control over the parameters of pitch and rhythm, respectively. It follows that other such devices may be utilized to increase control over other parameters. By repetition of parametric values on an instrument, musicians also develop their capacity to hear the discrete steps of a given parameter (Snyder, 2000). The more this is done with the specific acoustic measurements defined in Part One, the more likely it is that these parameters can be used effectively in actual compositions/ improvisations.

In practice, the procedure for attempting to train the ear to hear (and muscles to play) the values in an uncommon parameter is multi-faceted. First, it must be determined that the desired parameter can be negotiated by the particular instrument being used (the drum set may not be the best instrument to negotiate the fine gradations of a particular tuning system, for example). Next, an appropriate measurement device must be found or built that is capable of showing the value of the parameter being studied in real time. After this is done, it is also necessary to determine the useful values that will be aimed for within the parameter. For example, this may mean the development of a distinct tuning system or scale of brightness values. Once all of these criteria are met, then it is possible to get down to the real work of training one's ear to hear the gradations of the parameter, which will hopefully lead to the more creative and frequent use of these in musical forms.

Since first developing an interest in negotiating parameters in unique ways, I have found a number of concrete means of applying the procedure described in the last paragraph to my own aural and technical development. One of these involves training myself to hear the pitches of tuning systems other than Equal Temperament. What is needed for this task is a tuner that reads frequency numbers rather than letter names, as well as an instrument that can access continuous regions of the pitch spectrum. In my case, the slide trombone was used. Once this is found, however, it is possible to calculate the frequencies of a tuning system (I have worked with Just Intonation thus far) and then begin the process of committing the sound of these to memory. While one is limited by the capacity of one's memory in distinguishing subtle changes in pitch (rather than hearing them as out-of-tune pitches in another system), it is also true that increasing numbers of musicians are able to play in more than one tuning system (Snyder, 2000; Stone, 1980).

If using the visual cues of a measurement device is effective in teaching one to hear and play with uncommon tunings, it may also be advantageous to approach ear training from an opposite angle, by synthesizing electronic sounds with exact parametric values and attempting to imitate these on an instrument. I also used this technique to effect in studying alternate tuning systems and developing the powers of pitch discrimination. However, this technique is problematic when the addition of the two simultaneous sounds affects the overall perception of both in the parameter being investigated. The masking effects of prominent overtones from one sound may obscure the loudness or spectral content of another sound, so imitating synthesized sounds in these parameters may not help in training the ear to hear their variation. One other parameter where it has been useful to synthesize exact values to develop these by imitation is rhythm. Many audio editing software programs that can be used to generate the pitches in a tuning system, as described above, are also capable of synthesizing sounds of exact duration and placement in time. By this process it is possible to hear and play along with virtually any discrete subdivided pulse in relation to the overall tempo beats, thus developing facility in playing these polyrhythms. To this end even a metronome, which emits a single pulse within a certain musically defined range, can be useful, although it is perhaps more effective to be able to hear multiple pulses simultaneously and to set up their relationship exactly.

Of the parametric measurements described in Part One, those most difficult to develop by ear training are certainly the secondary parameters such as loudness, sharpness, timbral width, etc. This is largely due to the fact that devices to meter the values in these parameters in real time aren't readily available, although it certainly is possible to hear the effect of changes in these parameters. It is not clear, though, that many fine gradations of value in these parameters are perceptible, and this may further complicate the issue. However, in attempting to develop methods of hearing and executing changes in the secondary parameters, a tool was found that if useful in measuring loudness: the VU meter, common on many home audio components. What this does is measure the relative amplitude of sounds being received by a microphone and represents these graphically as a single value (or two, with the right and left channels of a stereo mic). Before use in studying loudness of musical sounds, the VU meter must be calibrated by testing what level of the meter corresponds to specific values on a particular loudness scale. If a 1000 Hz test tone is used in calibration, then the musically useful phons scale can be worked with easily. We saw in Part One that the loudness in phons of a particular sound is equal to that of a 1000 Hz tone of the same loudness in decibels. It follows that if one knows that the decibel level of a thousand Hz tone corresponds to a specific point on a VU meter (which can easily be determined by a sound level meter), then any musical sound that reaches that same point would have a loudness of the same number of phons.

13. Composition

The above sections of Part Three describe some of the unique ways that Sonogrammatics can be applied to the analysis and performance of instrumental music, but the concepts proposed in this theory are perhaps best utilized as tools for the composition/ improvisation of musical works. This is one area where it is possible to fully utilize all of the concepts suggested thus far, from the precise definitions of sonic parameters and their extents and limits to the application of musical operations that vary these parameters to create material for musical forms. Because of the endless possibilities of the combination of these elements, only a few examples of compositions will be examined in this last section. The discussion below is intended to illustrate the process of putting these together. In essence this was accomplished by combining compositional approaches to create a “sonogrammatic” music of rigorous variation within the major sound parameters.

Solo Music

After first deriving some of the basic concepts of Sonogrammatics, I decided to try my hand at creating musical works by applying various operations to the parametric values available to me on the slide trombone. To this end, I began a book of solo compositions, now numbering ten pieces, each exploring a different aspect of formal development while being connected to the set of compositions by a common idea. This idea was to explore the most fundamental distinction of sound types in Sonogrammatic theory: the difference between constant and gradient sounds and what this means in each of the major sound parameters. Although the rigorous categorization of the parametric values of the trombone wasn't possible due to the lack of a database, I decided it was possible to begin working with applying operations to the verbal descriptions of these values. The resulting music is certainly unique, not owing allegiance to any particular musical style or idiom, and indicates the first successful steps toward using Sonogrammatics as a complete and holistic set of musical tools.

A fundamentally new way of thinking about music often requires a fundamentally different style of notation. For this reason the first four pieces examined here were written in a style developed expressly for the use of Sonogrammatic concepts. The appearance of this notational style resembles that of a flowchart, with a set of five rows each representing the progression of values within a given parameter. From the top down, the rows indicate values of pitch, duration, temporal envelope, spectrum and loudness. From the standpoint of visualizing parametric operations this notation is ideal; practically, however, like with any "graphic" notation, one must get used to deciphering the symbols and moving from one set of these to the next. This process hinders the ability of a trained musician to sight read the material, although this deficiency may diminish with experience. The advantage of being able to see the form of a piece laid out in clear terms by this notational style may make up for this fact.

In my own experience, the mastery of these pieces, whether traditionally notated or written as a flowchart takes time, mostly due to the mental juggling needed to perform multiple parameter changes from sound to sound. In some of the pieces, a rigorous compositional design was plotted beforehand, and certain passages written in the development of these pieces were subsequently cut out of the final version due to the sheer impossibility of sounding some of the parameter combinations. On the other hand, combinations that normally wouldn't get used in my music became integrated into my overall performance style. The task of learning each aspect of each sound, one by one, until an entire piece was ready for performance has greatly increased my awareness of the parameters of sound and their negotiation.

"Sisso"

The first piece that we will look at here, titled "Sisso" (**Score #1**), was designed so that a maximum of variation occurs from one sound to the next in both gradient and constant parameters. All of the sounds are gradient in either the pitch, spectrum, loudness, or temporal envelope (rate of speed) parameter, and the pattern of gradient parameters is such that all of parameters are gradient before any becomes so again. This serial arrangement of the gradient/ constant parameters is reflected in the serialization of the other parameters as well, although no attempt was made to unify the changes in one parameter with that of any other. The durations of every successive twelve sounds follow the integer model of a different twelve tone row in seconds, for example, as do the starting pitches of these sounds. The form of this piece is basically one long section, and, due to the combinations of operations across all of the parameters, the sound of the music is highly unpredictable.

"Repat 1" & "Repat 2"

In contrast to the distinctly non-repetitive form of "Sisso", the next two scores work to utilize the concept of repetition in different ways. In "Repat 1" (**Score #2**), in addition to the duration (in seconds) indicated in the second row, another value denotes the number of times that that sound is to be repeated. Again every sound has one gradient parameter, and for this piece the parameter of duration can be gradient as well, since the repetition of sounds means that their successive durations can be manipulated in an increasing or decreasing fashion. The form of this piece is a quasi-additive process, where the values in a parameter remain the same until that parameter becomes gradient. The results of this process sound as if the sounds in this piece slowly morph over time. The form of "Repat 2" (**Score #3**) is the same as that of "Repat 1" except for the crucial difference that the values of the parameters in the former piece are left to performer to choose from a menu of options. Still only one parameter at a time is changed, and the music does gradually change as well, but the factor of improvisation in this rule-based structure radically changes the timing of motion from event to event. More than one version of this piece has been performed at concerts, and each had a similar sound, although the number of repetitions of each sound and their order gave each version a distinctive feel.

"Theedie"

This piece (**Score #4**) was an attempt at developing a "theme" that consisted of a group of four sound events (the first four of the piece) by inversion, retrograde and transposition of both the individual gradient parameters of each sound and the ordering of the sound events. By repeating the theme while simultaneously changing the values of some of the parameters within certain sounds, this sound of this piece lies somewhere in between the highly repetitive forms of "Repat 1 & 2" and the non-repetitive "Sisso". Because most of the parameters change from one sound event to the next, there is a sense of motion or progression, but the recursion of the groups of sounds adds to the memorability of the music as a whole. At the end of the piece the development of the theme comes full circle in that the gradient parameters of the theme and the order of its events are presented in retrograde form. Of the solo pieces analyzed in this section, "Theedie" is perhaps the most effective at illustrating the effect of multiple operations on the parameters of sound.

“Iffle 2”

The last solo trombone piece that we will analyze, the second of two pieces with the title “Iffle” (**Score #5**), was written using the conventions of musical notation common in Western music. While this was done in order to speed up the rehearsal process, it also shows that the same amount of detail that went into the previous pieces can be included in a traditional score. The form and parametric operations of this piece, however, are much less visually distinct than in the Sonogrammatic notation of “Theedie” or “Sisso.” In any case the method used in “Iffle 2” was that of integral serialization, where all of the parameters are unified, such that each of the twelve pitch classes has a unique duration, loudness, and timbre and these are the same for every recurrence of that pitch throughout the composition. As with “Theedie,” this piece has repetitive elements yet is varied enough that a momentum is built from event to event. Interestingly, the sequence of sounds, whose order is the only parameter operated on in this piece although there is a progression of values in the other parameters, is long enough that individual sounds escape short-term memory. For this reason, the sound of “Iffle 2” is akin to “Sisso” in that the progression of sounds appears to be unpredictable.

Ensemble music

Since the forms of the above examples generally fall under the category of single process music (even if the process happens to be a combination of operations), it would be beneficial to analyze a composition that is distinctly section, for contrast. Moreover, in doing this we can introduce the idea of the creation of ensemble music by the methods laid out in Sonogrammatic theory. Although the full-scale acoustical analysis of simultaneous sounds creates a wealth of new issues in terms of negotiating sonic parameters, the examination of the individual instrumental parts of a musical work and how they relate to each other sets the stage for further research.

"Velkaw"

This piece for trio of trombone, contrabass and drum set (the drum set part is improvised according to verbal directions, see **Score #6**) was written with the purpose of using various parametric operations in distinct sections that each convey a different musical process. Much, if not all, of the pitch material was developed by applying pc-space operations to a group of pitch sets, and in fact many distinct phrases can be discerned that are repetitions of a set varied by octave displacement. The rhythms of the entire work were also generated by transformational operations applied to the integers of the pitch sets. In some cases, the pitch and duration parameters have a 1:1 relationship, while in other sections this is not the case. Secondary parameters in the melodic sections of this composition generally change at the phrasal level in order to retain the integrity of grouping these phrases into sections. In addition to the above, there are also various open spaces throughout the piece, such as at rehearsal letter H, where pitches are improvised on top of the parametric operations of loudness and duration.

In terms of the overall musical flow of this piece, contrast between sections is generally indicated by the relationship of one instrumental part to the other; there are sections where the notated parts alternate phrases, play in rhythmic unison, or develop independently of each other. The strategic manipulation of relative unison/ disunion is certainly one of the new parameters that must be accounted for in ensemble as opposed to solo music, as is the texture that these simultaneous processes create. Material can also be repeated with new orchestration, as seen beginning at rehearsal letters F and J, to affect timbre change without the use of extended techniques. In spite of the fact that the separate instrumental parts can be analyzed individually according to Sonogrammatic theory, the piece was conceived as a whole, with the linear progression of musical processes proceeding across both parts. As a sectional piece, it is difficult to speak of the overall sound of "Velkaw," but in general there is a sense of the isolation of the parameters of rhythm, pitch and loudness, such that changes on the formal level in these parameters contribute to the sense of musical progression.

14. Conclusion

From the very beginning of this essay to this concluding section, we have seen the progression of successive levels of analysis of the processes involved in musical creation. In Part One we first defined the various important parameters of sound and how these are negotiated on acoustic musical instruments. From there it was possible to determine the capacity of manipulation of these parameters by a set of musical operations. This and the rules by which musical material generated by such operations is arranged into cohesive forms of composition and improvisation were the topics of Part Two. Part Three surveyed the practical musical applications of the concepts proposed in the first two parts. There is much work to be done in order to develop these concepts into more rigorous systems of creative music making, and it is possible to imagine an evolutionary expansion of this text as new models of parametric analysis and different generative and transformational operations are discovered. The present discussion is also limited by being applicable, for the most part, only with single sounds, and the expansion of the ideas presented here into complex ensemble situations is also inevitable. This introduction to the theory of Sonogrammatics, then, attempts to be comprehensive but not complete, and as such suggests an open-ended approach to musical creation that is accessible and enticing to a wide range of musicians to develop further.

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<u>Score #1: "Sisso"</u>

Sisso

<table>
<tr><td>G3</td><td>Ab3 < F3</td><td>F#2</td><td>C4</td></tr>
<tr><td>1"</td><td>11"</td><td>6"</td><td>4"</td></tr>
<tr><td>Ø | FM: 5/1</td><td>Ø</td><td>Ø</td><td>tremolo</td></tr>
<tr><td>Ø</td><td>closed</td><td>Ø > split tone</td><td>Ø</td></tr>
<tr><td>mp</td><td>f</td><td>p</td><td>ff > pp</td></tr>
<tr><td></td><td></td><td></td><td></td></tr>
<tr><td>D3</td><td>F4 > C4</td><td>B3</td><td>( )</td></tr>
<tr><td>10"</td><td>12"</td><td>9"</td><td>3"</td></tr>
<tr><td>Ø | tr: 6/1</td><td>Ø</td><td>Ø | AM: 5/1 > 3/1</td><td>Ø</td></tr>
<tr><td>Ø > closed</td><td>Ø</td><td>Ø</td><td>breath</td></tr>
<tr><td>pp</td><td>mf</td><td>f</td><td>p < f</td></tr>
<tr><td></td><td></td><td></td><td></td></tr>
<tr><td>E3</td><td>( )</td><td>Bb2</td><td>A3 > F#3</td></tr>
<tr><td>5"</td><td>2"</td><td>7"</td><td>8"</td></tr>
<tr><td>Ø | FM: 4/1</td><td>Ø</td><td>Ø | tr: 1/1 < 6/1</td><td>flutter</td></tr>
<tr><td>overtones < Ø</td><td>noise</td><td>Ø</td><td>Ø</td></tr>
<tr><td>ff</td><td>pp < mf</td><td>mf</td><td>mp</td></tr>
<tr><td></td><td></td><td></td><td></td></tr>
<tr><td>D4</td><td>A3</td><td>G2</td><td>E4 < F4</td></tr>
<tr><td>2"</td><td>12"</td><td>10"</td><td>1"</td></tr>
<tr><td>Ø | AM: 5/1 < 6/1</td><td>Ø</td><td>Ø</td><td>tremolo</td></tr>
<tr><td>Ø</td><td>closed</td><td>split tone < Ø</td><td>Ø</td></tr>
<tr><td>p</td><td>ff > mp</td><td>pp</td><td>mf</td></tr>
<tr><td></td><td></td><td></td><td></td></tr>
<tr><td>C3 < F3</td><td>B2</td><td>F4</td><td>F#3</td></tr>
<tr><td>5"</td><td>8"</td><td>4"</td><td>6"</td></tr>
<tr><td>Ø</td><td>Ø | tr: 3/1</td><td>Ø</td><td>Ø | FM: 6/1 > 1/1</td></tr>
<tr><td>split tone</td><td>Ø > closed</td><td>Ø</td><td>Ø</td></tr>
<tr><td>mp</td><td>f</td><td>ff > p</td><td>pp</td></tr>
<tr><td></td><td></td><td></td><td></td></tr>
<tr><td>Bb4 > G4</td><td>Db4</td><td>( )</td><td>Ab2</td></tr>
<tr><td>3"</td><td>9"</td><td>7"</td><td>11"</td></tr>
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<tr><td>Ø</td><td>closed</td><td>breath</td><td>closed > overtones</td></tr>
<tr><td>mf</td><td>mp</td><td>f > pp</td><td>p</td></tr>
<tr><td></td><td></td><td></td><td></td></tr>
<tr><td>( )</td><td>G3</td><td>Eb4 > C4</td><td>Bb3</td></tr>
<tr><td>3"</td><td>11"</td><td>6"</td><td>9"</td></tr>
<tr><td>Ø</td><td>Ø | AM: 7/2 > 1/2</td><td>Ø | AM: 4/1</td><td>Ø</td></tr>
<tr><td>noise</td><td>Ø</td><td>Ø</td><td>closed < Ø</td></tr>
<tr><td>mf < ff</td><td>mp</td><td>f</td><td>p</td></tr>
<tr><td></td><td></td><td></td><td></td></tr>
<tr><td>B1</td><td>F2</td><td>A3 > G3</td><td>C#3</td></tr>
<tr><td>2"</td><td>1"</td><td>8"</td><td>7"</td></tr>
<tr><td>Ø</td><td>Ø</td><td>flutter</td><td>Ø | FM: 6/1 > 1/1</td></tr>
<tr><td>Ø</td><td>Ø > closed</td><td>Ø</td><td>Ø</td></tr>
<tr><td>ff > mf</td><td>pp</td><td>p</td><td>mp</td></tr>
</table>

E2 < Bb2	F#3	C4	D3
12"	5"	4"	10"
Ø | slide ~	Ø	Ø | tr: 3/1 < 6/1	Ø
Ø	split tone	Ø	Ø > split tone
mf	f > p	ff	pp
( )	E4	F#4	C4 > Bb3
4"	10"	3"	7"
Ø | FM: 5/1	Ø | AM: 5/1 > 1/1	Ø	Ø | AM: 5/1
noise	closed	closed < Ø	Ø
mp < f	mf	f	ff
D4	Eb3	F2 < Ab2	( )
8"	2"	6"	1"
Ø | FM: 3/2 < 7/2	Ø	Ø | tr: 8/1	Ø
Ø	Ø < breath	Ø	breath
pp	p	mf	f > mp
A3	B2 < D#3	Ab2	Bb3
9"	5"	12"	11"
Ø	Ø	Ø | tr: 6/1	Ø | FM: 5/1 > 3/1
closed < Ø	Ø	Ø	Ø
ff	pp	p < f	mp
F#4	C2	B3 > Ab3	D#3
5"	2"	4"	10"
Ø | tr: 2/1 < 6/1	Ø	Ø	Ø
Ø	Ø	Ø	split tone < Ø
f	ff > mp	pp	p
Bb2	E3 > C3	Ab3	A2
12"	7"	9"	11"
flutter	tremolo	Ø | slide ~	Ø | AM: 1/2 < 7/2
Ø	Ø	Ø > split tone	Ø
mp > pp	mf	ff	pp
F2 > E2	C#3	( )	G4
1"	3"	6"	8"
Ø	Ø	Ø | tr: 6/1 > 1/1	Ø | slide ~
Ø	split tone < closed	breath	Ø
p	mp	mf	f > p
B2	G2	( )	F3 > Db3
6"	11"	7"	8"
Ø	Ø | FM: 3/2 < 5/1	Ø	Ø
Ø > split tone	overtones	noise	Ø
pp	p	mp < f	mf

E3 > C3	A3	Bb2	Eb2
3"	9"	5"	2"
tremolo	Ø | tr: 6/1 > 4/1	Ø	tremolo
Ø	closed	split tone < Ø	Ø
f	ff	mp	f < ff
D4 < E4	Db3	Gb3	C4
4"	10"	12"	1"
Ø | tr: 4/1	Ø | FM: 5/1	Ø	Ø | FM: 5/1
Ø	Ø	Ø > split tone	Ø
p	ff > pp	pp	mf
E3	D4	C4 > Ab3	A4
12"	1"	11"	2"
Ø	flutter	Ø	Ø | AM: 5/1 < 6/1
overtones < Ø	Ø	closed	Ø
f	p < mf	ff	pp
Db3	Bb2	Gb2 < Bb2	( )
10"	3"	9"	4"
Ø	Ø | tr: 4/1 > 2/1	Ø	Ø
split tone < closed	Ø	Ø	noise
mf	mp	p	ff > p
B2	F2	Eb2	G4 > E4
8"	5"	7"	6"
tremolo	Ø	Ø | tr: 6/1 > 1/1	Ø
Ø	Ø > overtones	Ø	Ø
pp < f	mf	mp	f
A3	G2	G#3	F#3 > E3
11"	6"	4"	10"
Ø | FM: 1/2 < 7/2	Ø | slide ~	Ø | FM: 6/1	flutter
Ø	Ø > split tone	Ø	Ø
ff	pp	mf < ff	mf
C4	D3 < F3	F4	B3
12"	9"	3"	5"
Ø | AM: 1/1 < 6/1	Ø	Ø | tr: 6/1	Ø
Ø	closed	Ø	closed < Ø
f	p	pp < ff	mf
C#2	E2 < A2	Eb4	Bb2
2"	7"	8"	1"
Ø	Ø | FM: 4/1	Ø | tr: 4/1 > 1/1	Ø
Ø	Ø	Ø > closed	Ø > closed
mp < ff	f	p	ff

Score #2: “Repat 1”

Repat 1

Gb2	Gb2 < Bb2	Bb2	Bb2
4" | 5X	4" | 7X	4" | 2X	4" | 6X
flutter	flutter	flutter < Ø	Ø
ah	ah	ah	ah > split tone
f > p	p	p	p
Bb2	Bb2	Bb2 > A2	A2
4" > .5"	.5" | 12X	.5" | 7X	.5" | 6X
Ø	Ø	Ø	Ø | FM: 6/1 > 5/1
split tone	split tone	split tone	split tone
p	p < mf	mf	mf
A2	A2	A2	A2 < C#3
.5" | 8X	.5" < 3"	3" | 3X	3" | 5X
Ø | FM: 5/1	Ø | FM: 5/1	Ø | FM: 5/1	Ø | FM: 5/1
split tone < overtones	overtones	overtones	overtones
mf	mf	mf < ff	ff
C#3	C#3	C#3	C#3
3" | 2X	3" | 4X	3" < 7"	7" | 1X
Ø | FM: 5/1 > Ø	Ø	Ø	Ø
overtones	overtones < ah	ah	ah
ff	ff	ff	ff > mp
C#3 < E3	E3	E3	E3
7" | 6X	7" | 4X	7" | 3X	7" > 2"
Ø	Ø | tr: 1/1 < 6/1	Ø | tr: 6/1	Ø | tr: 6/1
ah	ah	ah > closed	closed
mp	mp	mp	mp
E3	E3 < G3	G3	G3
2" | 5X	2" | 7X	2" | 12X	2" | 4X
Ø | tr: 6/1	Ø | tr: 6/1	Ø | AM: 5/1 > 4/1	Ø | AM: 4/1
closed	closed	closed	closed > Ø
mp > pp	pp	pp	pp
G3	G3	G3 < C#4	C#4
2" < 6"	6" | 9X	6" | 3X	6" | 4X
Ø | AM: 4/1	Ø | AM: 4/1	Ø | AM: 4/1	Ø | AM: 4/1 > Ø
Ø	Ø	Ø	Ø
pp	pp < f	f	f
C#4	C#4	C#4	C#4 < E4
6" | 7X	6" > 10"	10" | 2X	10" | 3X
Ø	Ø	Ø	Ø
Ø > split tone	split tone	split tone	split tone
f	f	f > mp	mp

<u>Score #3: “Repat 2”</u>

Repat 2

E2 < G#2	2 "	-- | AM: 4/1 > 3/1	ah	mf > p
Bb2 > G2	5 "	Ø	split tone	f
C4 > A3	3 "	^ | slide ~	closed	pp < mf
B3 < D4	6 "	tremolo	overtones	p
F4 > B3	9 "	- | tr: 5/1	breath	mf < ff
B2 < F3	4 "	Ø | FM: 6/1 > 2/1	noise	f > pp

Score #4: "Theedie"

Theedie

F3 > Db3	B3	Eb3	F2
5 "	3 "	6 "	2 "
Ø	Ø | tr: 4/1 < 6/1	Ø	Ø
ah	closed	Ø	ah > overtones
mp	mf	p < f	f
F3 > Db3	B3	Eb3	F2
5 "	3 "	6 "	2 "
Ø	Ø | tr: 4/1 < 6/1	Ø	Ø
ah	closed	Ø	ah > overtones
mp	mf	p < f	f
F3 > Db3	B3	Eb3	F2
5 "	3 "	6 "	2 "
Ø	Ø | tr: 4/1 < 6/1	Ø	Ø
ah	closed	Ø	ah > overtones
mp	mf	p < f	f
F3 > Db3	B3	( )	F2
5 "	3 "	6 "	2 "
Ø | FM: 3/1	Ø | tr: 4/1 < 6/1	Ø	Ø
ah	closed	noise	ah > overtones
mp	mf	p < f	f
F3 > Db3	C2	( )	F#3
5 "	3 "	6 "	2 "
Ø | FM: 3/1	Ø | tr: 4/1 < 6/1	Ø	Ø
ah	closed	noise	ah > overtones
mp	mf	p < f	f
F3 > Db3	C2	E3	F#3
5 "	3 "	6 "	2 "
tremolo	Ø | tr: 4/1 < 6/1	Ø	Ø
ah	closed	overtones	ah > overtones
mp	mf	p < f	f
F3 > Db3	D4	E3	A2
5 "	6 "	4 "	2 "
tremolo	Ø | tr: 4/1 < 6/1	Ø	Ø | FM: 5/1
ah	closed	overtones	ah > overtones
mp	mf	p < f	f
F3 > Db3	F#2	Bb3	E3
1 "	6 "	4 "	2 "
Ø	Ø | tr: 4/1 < 6/1	flutter	Ø | FM: 5/1
closed	closed	Ø	ah > overtones
mp	mf	p < f	f

F3 > Db3	( )	C3	B2
1 "	3 "	2 "	5 "
flutter	Ø | tr: 4/1 < 6/1	Ø	Ø | tr: 4/1
ah	breath	Ø	ah > overtones
mp	mf	p < f	f
F3 > Db3	D#4	Bb2	E2
4 "	6 "	2 "	1 "
Ø | tr: 4/1	Ø | tr: 4/1 < 6/1	flutter	Ø
ah	closed	Ø	ah > overtones
p	mf	p < f	f
F3 > Db3	Ab3	A2	F2
5 "	3 "	6 "	4 "
Ø	Ø | tr: 4/1 < 6/1	Ø | tr: 6/1	Ø | tr: 6/1
closed	closed	Ø	ah > overtones
f	pp	p < f	f
F3 > Db3	Db2	( )	G2
4 "	3 "	1 "	2 "
Ø | slide ~	Ø | tr: 4/1 < 6/1	Ø	Ø
ah	closed	noise	ah > overtones
ff	mf	p < f	f
D4 > Bb3	Eb4	G2	F4
5 "	6 "	1 "	4 "
Ø | FM: 3/1	Ø | tr: 4/1 < 6/1	Ø	Ø
ah	closed	ah > overtones	Ø
mp	p	f	p < f
B3	Ab2	A4	Ab3 > E3
2 "	3 "	6 "	5 "
flutter	Ø	Ø | tr: 4/1 < 6/1	tremolo
Ø	ah > overtones	Ø	ah
p < f	f	mf	f
Bb3	A2 > F2	E2	C#3
6 "	5 "	4 "	2 "
Ø | tr: 4/1 < 6/1	flutter	Ø | FM: 5/1	Ø
closed	ah	ah > overtones	overtones
p	mp	f	p < f
C4	D2	Eb2 > B1	F#2
6 "	2 "	1 "	6 "
flutter	Ø | AM: 5/1	Ø	Ø | tr: 4/1 < 6/1
Ø	ah > overtones	closed	Ø
p < f	f	mf	mp

Db3 < F3	C#4	Ab2	Bb2
2 "	4 "	5 "	3 "
Ø	flutter	Ø | tr: 6/1 < 2/1	Ø
Ø	ah	ah	ah > closed
mf	f > p	p	mf
D4	B2 > D#3	Eb3	B2
2 "	1 "	5 "	6 "
Ø | tr: 2/1 < 3/1	Ø	Ø | AM: 6/1	- | FM: 5/1
ah	Ø	closed	closed > split tone
f	ff	pp < mf	f
F#3	F4 > E4	F#2	C2
5 "	3 "	2 "	1 "
Ø | trill	tremolo | tr: 6/1	- | tr: 6/1	Ø | tr: 4/1
overtones < ah	ah	ah	ah
ff	mf	p < f	mp
D4	A3 < C4	Bb3	F#4
1 "	6 "	4 "	2 "
tremolo	Ø	Ø | tr: 6/1 > 3/1	Ø
ah	ah	overtones	ah > closed
ff > mp	mf	f	f
C3	A2	G4	Bb2 > E2
3 "	6 "	4 "	5 "
Ø | slide ~	Ø | tr: 1/1 < 5/1	Ø | FM	Ø | slide ~
overtones < closed	overtones	ah	ah
mf	ff	mp < mf	mf
Gb2	E3	Ab3 < A3	E4
6 "	4 "	2 "	5 "
Ø	Ø	Ø | AM: 5/1	Ø | tr: 4/1 < 8/1
overtones	ah > closed	closed	ah
f > p	f	mp	p
F3	( )	E2 < G2	Ab2
1 "	5 "	3 "	6 "
Ø | tr: 8/1	Ø	-- | flutter	Ø
closed	noise	ah	split tone > overtones
ff	mp < f	mf	ff
F2	B3	F3 > Db3	Eb3
2 "	3 "	5 "	6 "
Ø	Ø | tr: 4/1 < 6/1	Ø	Ø
ah > overtones	closed	ah	Ø
f	mf	mp	p < f

Bb3	F#4	D4	A3 < C4
4 "	2 "	1 "	6 "
Ø | tr: 6/1 > 3/1	Ø	tremolo	Ø
overtones	ah > closed	ah	ah
f	f	ff > mp	mf
F#2	C2	F#3	F4 > E4
2 "	1 "	5 "	3 "
- | tr: 6/1	Ø | tr: 4/1	Ø | trill	tremolo | tr: 6/1
ah	ah	overtones < ah	ah
p < f	mp	ff	mf
Ab2	Bb2	Db3 < F3	C#4
5 "	3 "	2 "	4 "
Ø | tr: 6/1 < 2/1	Ø	Ø	flutter
ah	ah > closed	Ø	ah
p	mf	mf	f > p
Eb3	B2	D4	B2 > D#3
5 "	6 "	2 "	1 "
Ø | AM: 6/1	- | FM: 5/1	Ø | tr: 2/1 < 3/1	Ø
closed	closed > split tone	ah	Ø
pp < mf	f	f	ff
Db3 < F3	( )	C3	B2
1 "	3 "	2 "	5 "
flutter	Ø | tr: 6/1 < 4/1	Ø	Ø | tr: 4/1
ah	breath	Ø	overtones < ah
mp	mf	f > p	f
Db3 < F3	D#4	Bb2	E2
4 "	6 "	2 "	1 "
Ø | tr: 4/1	Ø | tr: 6/1 < 4/1	flutter	Ø
ah	closed	Ø	overtones < ah
p	mf	f > p	f
Db3 < F3	Ab3	A2	F2
5 "	3 "	6 "	4 "
Ø	Ø | tr: 6/1 < 4/1	Ø | tr: 6/1	Ø | tr: 6/1
closed	closed	Ø	overtones < ah
f	pp	f > p	f
Db3 < F3	Db2	( )	G2
4 "	3 "	1 "	2 "
Ø | slide ~	Ø | tr: 6/1 < 4/1	Ø	Ø
ah	closed	noise	overtones < ah
ff	mf	f > p	f

Db3 < F3	B3	( )	F2
5 "	3 "	6 "	2 "
Ø | FM: 3/1	Ø | tr: 6/1 < 4/1	Ø	Ø
ah	closed	noise	overtones < ah
mp	mf	f > p	f
Db3 < F3	C2	( )	F#3
5 "	3 "	6 "	2 "
Ø | FM: 3/1	Ø | tr: 6/1 < 4/1	Ø	Ø
ah	closed	noise	overtones < ah
mp	mf	f > p	f
Db3 < F3	C2	E3	F#3
5 "	3 "	6 "	2 "
tremolo	Ø | tr: 6/1 < 4/1	Ø	Ø
ah	closed	overtones	overtones < ah
mp	mf	f > p	f
Db3 < F3	D4	E3	A2
5 "	6 "	4 "	2 "
tremolo	Ø | tr: 6/1 < 4/1	Ø	Ø | FM: 5/1
ah	closed	overtones	overtones < ah
mp	mf	f > p	f
Db3 < F3	F#2	Bb3	E3
1 "	6 "	4 "	2 "
Ø	Ø | tr: 6/1 < 4/1	flutter	Ø | FM: 5/1
closed	closed	Ø	overtones < ah
mp	mf	f > p	f
Db3 < F3	B3	Eb3	F2
5 "	3 "	6 "	2 "
Ø	Ø | tr: 6/1 < 4/1	Ø	Ø
ah	closed	Ø	overtones < ah
mp	mf	f < p	f
Db3 < F3	B3	Eb3	F2
5 "	3 "	6 "	2 "
Ø	Ø | tr: 6/1 < 4/1	Ø	Ø
ah	closed	Ø	overtones < ah
mp	mf	f < p	f
Db3 < F3	B3	Eb3	F2
5 "	3 "	6 "	2 "
Ø	Ø | tr: 6/1 < 4/1	Ø	Ø
ah	closed	Ø	overtones < ah
mp	mf	f < p	f

Score #5: "Iffle 2"

Score #6: "Velkaw"