DIFFERENCE THRESHOLDS FOR TIMBRE RELATED TO AMPLITUDE SPECTRA OF COMPLEX SOUNDS

by

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ABSTRACT

The sensation of timbre has intrigued and confused musicians and psychoacousticians for over a century. Current timbre definitions are not in agreement. Most are negative or catch-all in nature, and state that timbre is the sensation left when loudness and pitch are ignored.

This investigation was conducted to determine an individual's sensitivity to changes in timbre as a function of the intensity change of one partial in the spectrum of a complex sound. The relation of differential sensitivity to partial number and loudness also was investigated. The determination of a difference threshold for timbre related to the power spectra of complex sounds yielded a more precise definition for the term "timbre".

Six subjects were recruited from music camp participants and music students at The University of Kansas: Four subjects were music campers, one was an undergraduate music student, and one was a graduate music student. Each subject participated in one training session and six measurement sessions over two days.

A modified method of limits was employed to determine the difference threshold for timbre. The standard stimulus was a complex of seven in-phase simultaneous harmonic sinusoids with a fundamental of 500 Hz. The comparison stimuli had energy added or subtracted from a given partial of the standard and redistributed among the other six partials. There were three random orders for the pairs of standard and comparison stimuli. Each member of the stimulus pair was presented for two seconds, with one second of silence between members of the pair. Four

seconds of silence were allowed between simulus pairs of stimuli for the subject's response.

The stimuli were synthesized digitally using the MUSIC V program. Digital to analog conversion was at 17,500 samples/sec. sampling rate, and was recorded on standard recording tape. The tape was played through a preamplifier having a low-pass filter to help reduce switching transients. The stimuli were presented monaurally to the left ear at 70 db SPL.

During each of the six measurement sessions, each subject responded to seven sets of 19 stimulus pairs--one set for each partial varied in the seven-component complex tone. Three additional subjects as well as those participating in the timbre investigation listened to a tape of stimuli varying the third partial and evaluated loudness differences between stimulus pairs.

A computer recorded the subjects' "same" and "different" responses, and unscrambled the random stimulus order. The transitional points were the intensities of the partial varied in relation to the others, where the same response changed to different, with at least two responses in the new direction. The upper and lower thresholds were the midpoints between these transitional points and the next stimulus value in each direction.

Difference thresholds (expressed in db Calculated SPL) were subjected to a treatment by subject analysis of variance with partial varied as the treatment condition. The results indicated no significant difference between column means, supporting the hypothesis that there would be no significant differences in difference thresholds obtained by

varying each of the seven components of the standard stimulus. No significant difference was found for upper or lower threshold data subjected to treatment by subject analysis of variance. Therefore, an average difference threshold for timbre for the standard timulus was calculated, and found to be 4.28 db CSPL.

No subject gave consistent responses for loudness changes. It was concluded that, for the standard timulus, a larger power spectrum change was required to induce a change of loudness than to induce a change of timbre. Further, it was concluded that it was perfectly possible to determine a difference threshold for timbre, and that each component of the complex contributes equally to the overall timbre sensation. A new definition for timbre was forwarded: Timbre is that aspect of sound sensation related to the power spectrum of a complex when pitch and loudness are held constant.

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CHAPTER I

INTRODUCTION

Background

Music is sound and silence moving through time in perceptible forms expressive within a context (Heller, 1976). Inherent in this definition is the concept of perception, the awareness of objects or data through the medium of the senses. Perception, in a general sense, is concerned with an individual's understanding of the surrounding physical world. Furth (1970) explains that perception is a knowing activity that is focused on the immediately available sensory data. Perception refers to relations between the input to a biological system and the output of that system, both potentially observable. The task of the person studying these relationships is the "task of making inferences, or guesses, about relations" (Dember, 1960, p. 7). Thus perception is the catalyst for the interchange between organism and environment, and is the foundation for knowledge.

Music is concerned with auditory perception. Seashore (1938) states:

> Musical art and everyday experience of sound may proceed without any knowledge of physics, physiology, or psychology; but when the scientist attempts to explain these experiences he must deal with the series as a whole, the sound wave, the nerve impulse, and the experience of sound. The object of our study is music from the psychological point of view. Music is the center of our interest, the goal towards which we are working. (p. 15)

Music educators therefore must realize that knowledge of music is derived in part from perception. How an individual perceives auditory stimuli, and the limitations both physiologically and congnitively which govern his or her perception, are of primary concern to the music educator.

Physiological descriptions of auditory reception are now quite detailed (viz., Flanagan, 1972). The physics of sound, embraced in the discipline of acoustics, has been studied since the time of Pythagoras (Backus, 1969, p. xi). The study of living organisms, rather than "things," has had a comparatively brief history, although they both are concerned with behavior. The relation between sound's physical and perceptual aspects has been investigated largely in the last century.

The science of psychophysics developed in response to the need for understanding the relation between physical and psychological parameters. Roederer (1975) states that

> psychophysics tries to make predictions on the evolution of a specific system subjected to given initial conditions. The system under consideration is the brain and associated peripheral nervous and endocrine systems, the conditions are determined by the physical sensorial input stimuli, and the evolution is manifested by the individual physiological reactions or by the whole complex behavior of the body commanded by that brain. (p. 8)

The study of the relation between physical stimuli and organismic response was given a systematic approach by Fechner (1860) in his treatise Elements of Psychophysics. In the preface to this work, the goals of psychophysics are clearly stated, and remain valid today:

> By psychophysics . . . I mean a theory which . . . is new insofar as its formulation and treatment are concerned; in short it is an exact theory of the relation of body and mind.
>
> As an exact science, psychophysics, like physics, must rest on experience and mathematical connection of those empirical facts that demand a measure of what is experienced or, when such a

> measure is unavailable, a search for it. Since the measure of physical magnitudes is already known, the first and main tasks of this work will be to establish the as yet non-existant measure of physic magnitudes; the second will be to take up the applications and detailed arguments that develop from it. (p. xii)

Psychoacoustics is the psychophysical study of sound. It is a subtest, or branch, of psychophysics in general. In response to the challenge issued by Fechner almost 100 years ago, this research will investigate the psychoacoustics of timbre.

Sensation

Classical psychophysics makes a distinction between perception and sensation. The general consensus is that perception is more complex than sensation. The two are further distinguished in that "perception is relatively more dependent upon learning, motivational, and social, and personality factors than sensation" (Kimble, 1956, p. 143). Dember (1960) notes that such distinctions are suspect. The first basis for distinction, a tendency toward greater complexity, as well as the second criterion, dependence on other factors, are nebulous at best. Kimble (1956) seems to acknowledge this vagueness in stating: "As is apt to be the case with dichotomies, however, the sensation-perception one is somewhat arbitrary. The borderline between sensation and perception is often obscure" (p. 143). In this study the terms sensation and perception will be treated as synonymous.

A definition of sensation is difficult, but approachable. The "psychic magnitudes" mentioned by Fechner are clearly synonymous with the sensations spoken of by contemporary psychophysicists. Pavlov (1927) was one of the first experimenters to suggest a cortical mosaic upon which neural impulses from the sensory transducers are displayed. This same concept is found in Roederer's (1975) definition of

sensation: "Sensation is related to neural activity, evoked by sensory input signals, and presented in image on the cortical area associated with the specific sensory transducer" (p. 8).

Important to the concept of sensation is that of attention. Many sensory input signals may travel through the nervous system unattended by the biological system. In order for sensation to occur, the organism must be aware of these signals. In humans, this awareness is facilitated by the selective action of the reticular activating system of the lateral lemniscus. When this physiological system is stimulated by sensory input signals, sensation occurs.

Many physiologists agree that the location, amplitude, and spatial distribution of neural activity determine the class and subjective intensity of the associated sensation. Sensations can be classified into more or less well-defined types (Roederer, 1975). This fact is demonstrated by the many labels invented for their description, such as pitch, loudness, and timbre, even though there may not be mutual recognition of the meanings of those labels.

Another important aspect of sensation is that two sensations, one following the other, can be ordered by the individual as to whether a specific attribute of one is perceived as different from the other. The fact that such differences can have amount, that is, that there can be greater or lesser differences within a given set of sensations, indicates the possibility for measuring the magnitude of such differences, and hence the magnitudes, relative to a given criterion, of the individual sensations.

Sensation thus arises as a construct, "a conception built upon the objective operations of stimulation and reaction" (Stevens, 1975, p. 51). The study of sensation involves the study of organismic responses, responses which can be quantified objectively. It is not concerned with the study of undefinable mental material that would, by definition, defy objective tests. However, when investigators are dealing with human subjects, rather than physical events, they are concerned with the subjective.

Measurement

It is the subjective nature of the study of sensations which fuels a controversy concerning psychophysical measurement. It is important that this controversy be resolved and an adequate definition of measurement be stated.

Those concerned with fundamental approaches to measurement claim that sensation cannot be measured. One of the central features of fundamental measurement is the measurement of numerosity, the quantity of countable things (Stevens, 1975). Numbers representing such quantities can be subjected to standard algebraic transformations with no loss of validity. Helmholtz (1877) was the first to suggest that any operation must be mirrored by the mathematical laws of additivity. If they do not, the operations do not qualify as measurement. In the fundamental measurement concept, mathematics is the model upon which emperical results must be compared. Since sensations cannot be counted "like beans" (Stevens, 1975), the measurement of sensation is, by definition, impossible.

This view has been embraced by a large number of scholars, from the late nineteenth century to the present. The absence of something to count stimulated William James to reject the idea of sensation measurement. James believed that "the whole notion of measuring sensations remains, in short, mere mathematical speculation" (James, 1890, p. 539). Wundt (1890) felt that, although people have the ability to judge one sensation as stronger or weaker than another, they cannot say how much stronger or weaker. Stumpf (cited in James, 1890) also felt that "one sensation cannot be a multiple of another. If it could [sic.], we ought to be able to subtract the one from the other, and feel the remainder by itself" (p. 547). Savage (1970) continues to argue from this concept that we should "abandon the concept of psychological magnitude" (p. 408).

Stevens (1940) developed a new concept of measurement. In essence, he rejected the rigidity of the rule that empirical information must be subjected to the test of mathematical rules. He reasoned that since mathematics is a description, and empirical information describes, one should be able to form a union between empirics and mathematical models.

> We find, then, that in the business of measurement the number system of mathematics can provide the schema; experimental manipulations can provide the facts, the empirics. Properly combined, the empirics and the schema form a schemapiric union that results in useful measurement. (Stevens, 1975, p. 41)

Numbers, therefore, are simply representations of empirical operations. The outcome of the union is a scale, of which there are four types: nominal, ordinal, interval, and ratio. Each scale form applies different features of the number system, depending on the ways the

numbers can be altered and still retain all of the empirical information. Three scales, nominal, ordinal, and interval, do not allow for the addition of numbers to retain empirical information, yet they still provide procedures for measurement relative to their criteria and limitations.

Measurement thus can be defined as the assignment of numbers to objects or events according to any consistent rule, as long as this rule is not random assignment. By careful combination of empirics and the mathematical model, sensation can be measured. Stevens (1975) would argue that he has developed the means to measure sensations on the ratio scale, preserving the numerosity of fundamental measurement. But such measurement is only one form; people can measure anything according to consistent rules.

In summary, psychophysics is the study of sensation, the relation between physical stimuli and the response of a biological system. Psychophysics is concerned with the measure of this relation, and the application of a particular set of numbers to observed behavior with given initial conditions. Psychoacoustics is the branch of psychophysics which uses sound as the physical stimulus.

The relation between the physical components of a sound event and the psychological response to that event is the basis for the study of psychoacoustics. Sounds can be defined in terms of four physical parameters: frequency, amplitude, time and waveform. Each of these is associated with a psychological counterpart: pitch, loudness, duration and timbre. In psychoacoustical research, these parameters serve as variables in experimental designs.

Experiments in psychoacoustics, and psychophysics in general, are designed to answer several important quesitons which are directed at the input side of the perceptual system. The physical aspects of sound serve as input variables; the output is the recorded behavior of the system. These important questions are: 1. What general types of energy is the system capable of receiving? 2. For a given type of energy, what is the least amount of energy required for the system to be activated. 3. For a given type of energy, what is the smallest difference in amount of energy that the system can react to? 4. How does the system react to variations in the amount of energy received?

Need for the Study

To date these questions have been answered for amplitude, duration, and frequency, as is evidenced in the material presented in Chapter II. Timbre has received considerably less attention. Plomp (1970) states that "it is quite remarkable how little attention timbre, as contrasted with loudness and pitch, has received in hearing research" (p. 391).

The index to Backus's (1969) The Acoustical Foundations of Music has only one entry for timbre; he seems to have equated (or confused) the term with tone quality. In Hearing--Its Psychology and Physiology (Stevens, 1938), there are extensive chapters on loudness and pitch, however timbre is not even mentioned in the glossary. Winckel (1967) in Music, Sound and Sensation, treats both loudness and pitch, but completely ignores timbre, per se, instead referring to "tone

color". Even texts dealing specifically with psychoacoustics ignore timbre, such as Harris' (1974) Psychoacoustics.

Those who have addressed themselves to the subject have generally made comments that are, at best, nebulous. Licklider (cited in Plomp, 1970) discusses the attributes of complex sounds, but concludes that until careful scientific work has been done, it is impossible to say more about timbre than that it is a "multidimensional dimension." Zwicker and Feldtkeller (1967) also limited their remarks concerning timbre to its multidimensional nature.

Studies in hearing theory also have neglected the area of timbre perception. Von Bekesy (1963) does not include timbre in his list of the primary attributes of auditory sensation. Green (1976), in An Introduction to Hearing, goes into much detail on loudness and pitch discriminations of complex sounds, but does not cite similar studies dealing with timbre. Plomp (1970) concludes that "hardly any results of explicit experiments on timbre are available" (p. 392).

Definition of Timbre

Thus psychoacoustics has not addressed itself to answering the fundamental questions about timbre as it has to pitch and loudness. The reluctance to deal with timbre is exceeded only by the number of different definitions given for the term itself.

Helmholtz (1877) described timbre as "that peculiarity which distinguishes the musical tone of a violin from that of a flute. . . Every different quality of tone required a different form of vibration, but on the other hand it will also appear that different forms of vibration may correspond to the same quality. . ." (p. 10).

Seashore (1938) defines timbre as "that characteristic of a tone which depends upon its harmonic structure as modified by absolute pitch and total intensity" (p. 97). Seashore differentiates between tone quality and timbre when he states that "physically the timbre of a tone is a cross section of the tone quality" at a given moment in time (Seashore, 1938, p. 10).

The American Standards Association (cited in Plomp, 1970, p. 397) defines timbre as "that attribute sensation in terms of which a listener can judge that two steady-state complex tones having the same loudness and pitch, are dissimilar." Similarly, Webster (1966) defines timbre as "the characteristic quality of sound that distinguishes one voice or musical instrument from another or one vowel from another (p. 1525).

All of these definitions are rather broad in nature, and indicate that timbre is dependent on several parameters of sound including the spectral envelope and its change in time, periodic fluctuations of the amplitude or fundamental frequency, and whether the sound is tone or noise. These definitions all have a common characteristic: they are all virtually negative descriptions. Plomp (1970) states that "apparently the timbre concept is loaded rather negatively as the total of all aspects of sound sensation when loudness and pitch are left out of consideration" (p. 398). Similarly, Howe (1975) notes that

> The timbre or 'tone quality' of a musical instrument has been used to denote that property which enables a listener to identify the instrument. It is thus a 'bushel basket' or 'catchall' concept that has caused many difficulties. It is clear, though, that there are many distinct qualities subsumed under the term timbre. (p. 23)

If timbre is defined in the broadest sense, the five major parameters listed by Schouten (1968) must be accepted: tonal vs. noiselike

character, spectral envelope, time envelope, change, and acoustic prefix. If the definition is understood in a much stricter way, the more dynamic aspects should be excluded and timbre should be understood as being mainly related to sound spectrum.

Timbre is a multidimensional phenomenon. Of the attributes of sound already presented, pitch and loudness are both one-dimensional. Loudness differences, for example, are fully described by a single scale from faint to strong; pitch differences can be described by a scale from low to high. There is no such single one-dimensional scale for the comparison of the timbres of various sounds.

It is the multidimensional nature of timbre which accounts for confusion in its definition and research. The tone of an instrument may be placed on a bright-dull scale, but such a scale will not account for the diversity of auditory sensation of various complex tones. As early as 1890, Stumpf (1890) listed 20 semantic scales describing the timbre of complex tones.

The multidimensional nature of timbre is also evident in the mathematical formula defining a complex sound (Green, 1976):

$$\underline{p}(\underline{t}) = \sum_{n=1}^{m} \underline{a}_{\underline{n}} \sin(2\Pi nft + \phi_{\underline{n}}) \qquad (1)$$

where the variables are $\underline{p}$ the pressure, $\underline{n}$ the number of components, $\underline{a}$ the amplitude, $\underline{f}$ the frequency, and ϕ the phase cycle. The many degrees of freedom of this equation indicate the multidimensional nature of timbre. From this mathematical definition, we can see that timbre is determined by the amplitude spectrum $\underline{a}_1$, $\underline{a}_2$, . . . , $\underline{a}_{\underline{m}}$ and the phase

relation $\phi_1, \phi_2, \ldots, \phi_m$ of the successive harmonics. Timbre of a complex tone containing m harmonics depends on $2(m-1)$ parameters.

In examining this evidence, it is clear that a uniform distinction between timbre and quality of a sound is not made. At times these terms are used interchangeably; at other times rather specific distinctions are made. Roederer's (1973, 1975) Introduction to the Physics and Psychophysics of Music, for example, lists this sensation as tone quality in the first edition and timbre in the second.

In this study, timbre is defined as that attribute of sensation in terms of which a listener can judge that two steady complex tones having the same loudness, pitch, and duration are dissimilar. Timbre is the psychological aspect of sound related to waveform. Timbre is a dimension of tone quality. Quality also includes such aspects as periodicity of the amplitude envelope, transient partials, and individual partial amplitude envelopes. This narrow definition of timbre is reflected in the studies of Plomp (1970, 1976) and is derived from the simple mathematical formula of the complex sound (Equation 1).

Lack of Research

The dearth of studies on timbre is largely related to its multidimensional nature. We know much more today about both the production and the physical structure of complex tones, but little quantitative data are available on the perceptual differences between these tones.

Much of the work dealing with timbre perception has dealt with comparisons and classifications of sounds. Factor analytic methods have been used to reveal a cognitive classification of instrument types

into "woodwind," "brass," and "string," and a classification of the sound of these instruments into groups determined by the relative amplitude of the sound's partials (Wedin and Goude, 1972). These methods have also been used to derive factors said to correspond to such abstract qualities as "feminine," "masculine," and "loneliness" (Rahlfs, 1966), or volume and density as related to clarinet timbre (Jost, 1972).

Scaling techniques have been developed for the judgment of similarity of stimuli and interpretation of the cognitive distances in n-dimensional space between these stimuli (Plomp, 1970; Miller, 1976). Investigations of timbre on a non-comparative basis have been confined to the analysis of overtone structure for specific instruments (Backus, 1971; Freedman, 1967; Seashore, 1938).

The application of psychophysical methods in investigating timbre is virtually nonexistent. Although timbre is multidimensional, it should be possible to isolate a few dimensions, and investigate some of the basic questions cited as central to psychophysics. Specifically, questions 3 and 4 (p. 8) should be investigated, since the first two have received some attention (Plomp 1970, 1976). For timbre, what is the smallest difference in amount of energy to which the system can react? How does the system react to variations in amount of energy?

An investigation of the relationship between the parameters causing the timbre sensation is called for by the lack of attention this has received. Although several musicality tests and recognition studies have dealt with timbre, these studies have failed to investigate

differential sensitivity in relation to a specific variable. To the writer's knowledge, no study has systematically investigated the absolute or differential thresholds for timbre, even though much attention to these areas has been accorded pitch and loudness.

Research Questions

Several research questions thus arise: 1. What is the sensitivity of an individual to changes in timbre as a function of the intensity change of one of the partials in the overtone structure of a complex sound? 2. Is this sensitivity to timbre sensation the same regardless of which partial is varied? 3. How do loudness and timbre relate?

A study of the difference threshold for timbre taking into account every aspect of the multidimensional nature of the stimulus would be an enormous, if not impossible, undertaking. Essential to this study is the narrow definition of timbre presented earlier. Further, it should be possible to isolate a number of parameters and eliminate them or render them dependent on the power spectrum.

Although timbre is multidimensional, it has been evidenced in simple tones. Helmholtz (1877) recognized that tones without harmonics have a typical frequency dependent timbre. This concept has been experimentally verified by Engel (1886), Stumpf (1890), Grassmann (1887), von Wesendonk (1909), and Kohler (1911). Important to this study is the fact that timbre can be related to relatively few parameters, as long as the extraneous parameters are held constant.

Psychoacoustical experiments with electronically generated steady-state tones, of equal pitch and loudness but different overtone spectra and relationships among the harmonics, show that timbre sensations are controlled mainly by the power spectrum (Plomp, 1970). Phase changes, although perceptible, play only a secondary role. The perceptibility of phase changes appears to increase in the higher harmonics (Licklider, 1957). The writer therefore has chosen to describe the sensitivity of human subjects to changes in timbre as related to variations of individual partial intensities in complex sounds with in-phase spectral envelopes. As many dimensions of the stimulus as possible will be held constant; this will facilitate the use of classical psychophysical methodologies, although these will be modified for this study.

Definition of Terms

Before stating the hypothesis, it is necessary to define certain terms used in this study. Because of the multidimensional nature of the stimulus, many factors must be controlled. It is necessary that these factors be identified and defined.

Tone

In this study, a tone is defined as any sound event eliciting the three primary sensations of pitch, loudness, and timbre. The term "simple tone" stands for periodic sound waves that are sinusoidal. The term "complex tone" stands for a non-sinusoidal periodic sound wave that is the sum of sinusoidal components called partials. A "harmonic" is a component in integral multiple relation to the fundamental, which is the lowest component. The mathematical definition of a sinusoidal wave is given by the equation

$$p(t) = A\sin(2\Pi ft + \phi). \qquad (2)$$

where the variable p = the pressure; t = the time; A = the amplitude; f = the frequency; and ϕ = the phase angle (Green, 1976).

Frequency and Pitch

The frequency of the wave is the number of times the wave repeats itself per unit time. In this study, the standard terminology will be used with reference to frequency, Hz (Hertz), which is the European designation for the former American cycles-per-second. The period of the wave is defined as the time it takes a wave to execute one cycle. Mathematically this is $1/f$, where f is frequency in Hz.

The psychological correlation to frequency in a sinusoidal wave is pitch. The definition of pitch is that "subjective property of a sound that enables it to be compared to other sounds in terms of 'high' or 'low'" (Backus, 1969, p. 110). In this study, the concept of the mel scale as a psychophysical measurement of pitch is rejected (Howe, 1975; Radocy & Boyle, 1979).

Amplitude, Intensity, and Loudness

The amplitude of a wave is defined as the distance that the sound-producing body moves during vibration. The greater the distance from the point of equilibrium, the greater the amplitude. In this study "instantaneous amplitude" refers to the excursion distance at any instant in time, while "peak amplitude" refers to the greatest displacement achieved by the sound source. The root-mean-square amplitude is a statistical average of all amplitudes at all times. For a sine wave, the root-mean-square (RMS) amplitude is equal to the peak amplitude multiplied by .707 (Harris, 1974).

The physical strength of a sound wave is defined in this study in several distinct ways. In Chapter III, the relationship of these definitions will be explored further.

The power (I) of a sound wave is defined by the relation

$$I = p^2 / p_o c, \qquad (3)$$

where I is the power per square centimeter (w/cm^2) in the sound wave, p is the pressure in the wave measured in dynes per square centimeter, and the denominator is the characteristic impedance of the air (Green, 1976). For a standard temperature and atmospheric conditions, $p_o c = 40$ dyne sec/cm^3.

The intensity level (IL) of a sound is measured in decibels (db), and is defined in terms of power ratios:

$$IL\ (db) = 10 \log_{10} (I_1 / I_0). \qquad (4)$$

In the above equation, I_1 is a measured intensity in w/cm^2, and I_0 is a reference intensity of 10^{-16} w/cm^2 (10^{-12} w/m^2). The reference intensity is an approximated value for the threshold of hearing at 1,000 Hz for normal hearing young adults.

The sound pressure level (SPL) of a sound is also measured in decibels, and is defined in terms of pressure ratios:

$$SPL\ (db) = 20 \log_{10} (p_1 / p_0), \qquad (5)$$

where p_1 is an observed sound pressure and p_0 is a reference pressure of .0002 $dyne/cm^2$ or 2×10^{-5} $newtons/m^2$. This reference pressure is considered to be the threshold of hearing. A decibel is thus a logarithmic

ratio which compresses a large range of intensities or pressures into a less unwieldly and workable system.

The sensation level (SL) of a sound is expressed in db Sl and represents the number of decibels that the sound is above the audible threshold for a given individual. This is always used to refer to an individual only; db HTL refers to an individual's threshold relative to established norms.

Loudness level (LL) is another sensation measurement related to the intensity of a sound. The LL, expressed in phons, of any frequency of sine tone, is equivalent to the SPL (re .0002 dyne/cm^2) of a 1,000 Hz tone judged to be of equal loudness. The loudness of a given sine tone can also be measured in sones, with one sone being equivalent to the loudness of a 1/3 octave band of noise centered at 3, 150 Hz, at a level of 32 db SL. Since this reference can only be made for individuals (recall that SL is an individual measurement), one sone is set arbitrarily equal to 40 phons.

Phase

The phase of a periodic waveform is defined as that portion of a cycle which has elapsed at a given instant of time, relative to some arbitrary reference point. In this study the reference point will be the point of equilibrium. The time period required to complete one cycle can be represented as 360° along the time axis because of the mathematical relation between simple harmonic motion and circular functions. Thus the phase at any point during a cycle may vary between 0° and 360°. The phase angle or phase difference of a given sound event refers to the relative location of two periodic waveforms at a given instant in time.

Complex Tone

As stated earlier, a complex tone is defined as a periodic waveform consisting of the sum of sinusoidal waves. The sound spectrum of a complex tone is defined as the frequency by amplitude plot for such a tone. Sound spectrum is synonymous with power spectrum and amplitude spectrum.

The intensity (I) of a complex tone is defined as the sum of the intensities of the components of the tone, as expressed by the following equation:

$$I_{(total)} = I_1 + I_2 + \cdot \cdot \cdot I_n, \qquad (6)$$

where n is the nth component and I is expressed in w/cm^2. The components must be in phase (0° phase difference).

The sensation of loudness of a complex tone is less accurately defined, primarily due to the infancy of the investigation. As defined in this study, the loudness of a complex sound is computed by the procedure developed by Stevens (1961, 1972) and approved by ANSI. The complex sound intensity is measured in third-octave bands. The Loudness Index (LI) is found by comparison to a calculation nomograph (Harris, 1974). Then the following formula is applied:

$$L_{total} = LI_{largest} + F\ (LI_{sum\ of\ the\ remainder} - LI_{largest}). \qquad (7)$$

The constant F is dependent on the frequency band of the loudest component. This yields the total loudness in sones for a complex sound. Conversion to phons is also possible by using a conversion scale built into the calculation nomographs.

No single, concise definition of the pitch of complex sounds is feasible at this time, due to the large number of conflicting studies on the subject. Various parameters influence the pitch elicited by a complex waveform; these will be identified and discussed in Chapter II.

Thresholds

A threshold is defined in this study as the minimum amount of energy required for the accomplishment of a perceptual task at the probability criterion of .50. The absolute threshold is thus defined as the minimum stimulus which is capable of first eliciting a response half of the time. The concept of an absolute threshold for timbre must be rejected, except with reference to the absolute threshold for loudness. It is clear that since even simple stimuli elicit a particular frequency dependent timbre, the moment of perception (the point of absolute threshold) will similarly yield a timbre response. This conclusion is, however, conjecture, and focuses attention on a further area of study with complex sounds.

The difference threshold in this study is defined as the smallest increment or decrement in a parameter of a stimulus which results in a change in sensation for a given subject. Synonymous with the term "difference threshold" are the terms "difference limen" and the "just noticeable difference." The difference threshold is also a statistical value, and is that change in a physical parameter which elicits a response of being different from a comparison stimulus 50 per cent of the time.

Purpose and Hypotheses

The purpose of this study was to apply classical psychophysical techniques to the problem of determining the difference threshold for timbre as related to changes in the power spectrum of a complex sound. The complex stimuli used in this study consisted of seven partials. Each partial was increased or decreased from a standard intensity value, with the surplus energy redistributed among the other components. Difference thresholds were measured for the seven partials, which served as independent variables.

The answer the research questions posed earlier, a hypothesis suitable for investigative solution was stated:

> There will not be a significant difference among subjects for difference thresholds obtained by varying each of the seven components of a complex sound.

To better understand the nature of this hypothesis, and the procedure under which differential sensitivity to timbre will be described, it was necessary to review the literature dealing with psychoacoustical studies of both simple and complex tones.

CHAPTER II

RELATED LITERATURE

A review of the literature related to the study of timbre naturally must involve the presentation of work representative of its multidimensional nature. This presentation therefore will cover three distinct areas: 1. Research dealing with difference thresholds themselves, in which the results of the work are not as important as theoretical and methodological considerations. 2. Research into the loudness, pitch, and phase aspects arising from multicomponent stimuli. 3. Research into timbre perception.

Difference Thresholds and Psychophysical Methodology

Developmental Background

Differential sensitivity is the relation between the difference threshold for intensity and the intensity level of the stimulus. Weber (1834) discovered that two relatively heavy weights must differ by a greater amount than two relatively light weights for one weight to be perceived as heavier than the other. The size of the difference threshold was found to be a linear function of stimulus intensity, with the stimulus always having to be increased by a constant fraction of its value to be just noticeably different to an observer.

Weber (1834) found that different sense modalities did not provide the same constant fractional value. He was able to specify a

lawful relationship between the size of the difference threshold and stimulus intensity level. This relationship is known as Weber's law: the change in stimulus intensity that can just be discriminated is a constant fraction of the starting intensity of the stimulus($\Delta\phi = c\phi$). In this form, Weber's law is not found to fit emperical data at low stimulus intensities (Engen, 1971). Konig and Brodhum (1889) found that in brightness discrimination, the value of $\Delta\phi$ decreased as intensity increased and then became approximately constant for the higher intensity values. Riesz (1928) found that discrimination for sound intensity followed a similar pattern; the brightness discrimination was apparently better than loudness discrimination. It was apparent that a constant value must be added to Weber's law to account for low intensity deviation. The resultant law is $\Delta\phi = c\ (\phi + a)$ (Gescheider, 1976).

Fechner (1860) extracted the theoretical framework for psychophysics from Weber's work. His initial premise was that an arithmetic series of mental intensities might correspond to a geometric series of physical energies. Fechner further proposed that sensation magnitude could be quantified indirectly by relating the values of the physical scale to the corresponding values of the JND in sensation on the psychological scale. His central assumption was that all JNDs were equal increments in sensation magnitude regardless of the size of $\Delta\phi$.

Since a basic unit was established (JND), it was simply a matter of counting units to specify the amount of sensory magnitude. The intensity in physical units of a stimulus at absolute threshold was assumed to correspond to the zero point on the psychological scale of sensation magnitude. Fechner (1860) soon discovered that sensation

magnitude plotted against the logarithm of the stimulus intensity produced a linear function. By integration over a series of values, Fechner's law was derived: stimulus magnitude equals a constant times the log of the stimulus over threshold - $\chi = K \log \phi$.

However, the validity of Weber's law is suspect (Engen, 1971). Further, Fechner's law is based on the assumption that the JND is an equal increment in sensation at all levels of stimulus intensity. Stevens (1936) has shown that JNDs along the intensive dimension are unequal. Fechner's law is no longer considered an accurate statement of the relationship between stimulus intensity and sensation magnitude (Gescheider, 1976).

The central assumptions of the classical threshold theory are that fluctuations in threshold are random and that sensory dimensions are continuous. Infrequently obtained psychometric functions, where response probability increases from 0 to 1 as a linear function of stimulus magnitude, became the basis of the neural quantum theory, first made explicit by Stevens, Morgan, and Volkmann (1941). They derived a linear psychometric function from the assumption that discrimination occurs along a sensory dimension within the observer that is made up of small discrete (quantal) steps.

The first evidence in support of a quantal theory of discrimination was reported by von Bekesy (1930). Similar results in loudness and brightness were obtained by Stevens, Morgan, and Volkmann (1941). Stevens (1972a) reviewed the data from a dozen investigations carried out over a span of forty years. Some 140 step-like functions for auditory

loudness and pitch were reproduced in his paper as support for neural quantum theories.

However, it has been found that the conditions required to produce the linear curve are extreme. Miller (1947) reported that the stimulus must be very carefully controlled. When the standard and comparison stimuli were bursts of white noise, a normal ogive rather than a linear function was obtained. Stevens (1972a) reports that if the observer is unable to maintain a constant criterion during an experimental session the psychometric functions will tend to give ogives rather than straight lines. Miller and Garner (1944) found that if the size of the neural quantum changes within a session, the function will be an ogive. Neural quantum theory has been criticized on both methodological and theoretical grounds (viz., Corso 1956, 1973; Wright, 1974).

In recent years, the concept of thresholds for sensory stimuli has come under serious doubt. Early psychophysicists assumed a close connection between the verbal responses of an observer and the concurrent neurological changes in the sensory system caused by stimulation. They assumed that, in a well controlled psychophysical experiment, the probability of a particular response was entirely a function of the stimulus and the biological state of the system. Tanner and Swets (1954) have proposed that statistical decision theory and certain aspects of electronic signal detecting devices might be used to build a model closely approximating how people actually behave in detection situations. The model they developed is called the Theory of Signal Detection (TSD).

Green (1960) applied the TSD to auditory stimuli. Each subject was told that he would be given money for each correct response when a signal was presented against a background of Gaussian noise. The subject was penalized an amount for a false alarm, responding when no signal was present. He also was told there would be a signal on 10 percent of 300 trials. Points were plotted on a graph in which the abscissa was the probability of responding "yes" when noise alone was present, and the ordinate was the probability of responding "yes" when a signal plus noise was presented. The resulting curve was the receiver operator characteristic curve; this then was subjected to statistical analysis to determine the index of detectibility for the subject.

Parducci (1970) found that in comparative loudness discriminations, the proportion of loudness judgements was independent of the presentation probabilities. This suggests that there are limitations when applying the methods of the theory of signal detection to differential sensitivity measurements.

Gescheider (1976) feels that the classical threshold theory does provide a useful means of measuring sensation in terms of the amount of stimulus energy necessary to produce certain changes in the observer's behavior. Sensation is thus treated as a concept which must be defined in terms of stimulus-response relationships. The extent to which the threshold or value of the matching stimulus has been measured carefully under controlled conditions will determine the extent to which measurement can be used to infer the operation of the sensory processes within the observer.

Psychophysical methodology for quantifying the relations between physical and psychological dimensions is primarily concerned with presenting a stimulus to an observer and asking whether or not he/she perceives it. The variable state of the biological system implies that an observer presented with the same auditory event on several trials is likely to perceive on some trials and not on others. This concept is central to the theory of signal detection. Classical psychophysical methodologies also define the threshold in statistical terms. Typically, the threshold has been defined as the stimulus value which is perceptible 50 percent of the time. This is usually an average threshold, obtained over a number of observations and observers (Gescheider, 1976). Fechner (1860) developed three methods for threshold measurement: the methods of constant stimuli, limits, and adjustment.

Method of Constant Stimuli

The method of constant stimuli can be used to determine difference thresholds. This method was used in many early studies to produce the difference threshold for pitch (Shower and Biddulph, 1931; Stucker, 1908). The observer's task in this procedure was to examine pairs to stimuli and to judge which stimulus produced a sensation of greater magnitude. The standard stimulus (St) had a fixed value; the comparison stimulus (Co) was changed from trial to trial. It was sometimes greater than, sometimes less than, and sometimes equal to the value of the standard stimulus. Five, seven, or nine values of the comparison stimulus were used. These were separated by equal distances on the physical scale. Each of the comparison stimuli were paried

several times with the standard stimulus in a random sequence. The observers reported which stimulus had the greater sensory magnitude. This report was verbal in earlier studies (Stucker, 1908), but was changed to electrical indicators when electronic sound sources became the norm.

A psychometric function was charted in which the value of the comparison stimulus in physical units was plotted on a graph against the proportion of responses indicating greater sensory magnitude. The value of the comparison stimulus at the .5 proportion "greater" response level was known as the point of subjective equality, and represented the value of the comparison stimulus which was subjectively equal to the standard stimulus. The upper difference threshold was the stimulus range from the point of subjective equality to the .75 point. The lower difference threshold was the stimulus range from the point of subjective equality to the .25 point. These two difference thresholds were averaged to give one difference threshold representative of a particular standard stimulus value.

Gescheider (1976) expressed concern over two possible sources of error in the method of constant stimuli. A space error may occur when comparison stimuli are presented to different receptors. Most auditory studies use only one receptor and are therefore void of this error. A time error can occur when the standard and comparison stimuli are presented successively, as in the case with auditory experiments. If the comparison stimulus is presented after the standard, the proportion of times it is judged greater is higher than when it is presented first. The most likely explanation for this is memory image fading.

This is not solved by placing the stimuli in extremely close proximity since the sensation images become confused. To counterbalance for time error effects, most method of constant stimuli studies present the standard stimulus first on half of the trials and second on the other half of the trials. The problem of the order of presentation of stimuli plays an important role in the controversy over procedure as related to the analytical power of the ear and lateral suppression; this will be evident in the material presented in the second section of this chapter.

The method of constant stimuli cannot be applied to the problem of determining the difference threshold for timbre. Since the method of constant stimuli is based upon magnitude as well as direction of the stimuli, its application is limited to stimuli with such characteristics. It is doubtful whether an observer can tell which of two stimuli has greater timbre; he can only determine if the two stimuli have the same or different timbre. The writer must reject this method for use in the present study.

Method of Adjustment

The method of adjustment has been used in many studies to determine the difference threshold for loudness. Reisz (1928), Feldtkeller and Zwicker (1956), and Harris (1963) have used variations of this method to confirm the difference threshold for loudness using modulated sine-tones. The observers'task was to adjust a comparison stimulus until it seemed equal to a standard stimulus. This procedure is also known as the method of average error, since the experimenter is concerned with the magnitude of the discrepancy between the adjusted comparison stimulus and the fixed standard (Hays, 1967). Over a large

number of trials, the settings of the observers were sometimes less than and sometimes greater than the standard stimulus. It was assumed that a frequency distribution of the results would be more or less symmetrical, and if enough trials were held, would be distributed normally. The mean of this distribution was defined as the point of subjective equality. The difference threshold was the amount of dispersion in the settings, and thus emerged as the standard deviation of the distribution. If the standard deviation of the distribution was large, this indicated that over a wide range of stimulus values the two stimuli appeared equal and discrimination was poor. If the standard deviation was small, this indicated that the observations were clustered around the value of the standard stimulus and that discrimination was good.

The method of adjustment is difficult to apply when stimuli are not continuously variable. The precise control of the parameters of a complex tone is very difficult, therefore this method does not lend itself for use in determining the difference threshold for timbre.

Method of Limits

The most popular method for determining the difference threshold is the method of limits (Gescheider, 1976). Bekesy (1967) and Gassler (1954) used the method for measuring the difference threshold and absolute threshold for loudness. The method was used to investigate differential sensitivity for loudness in relation to stimulus duration by Harris (1963). Turnbull (1944) determined the difference threshold for pitch in relation to duration using the method of limits.

In a typical experiment, the observer was presented with a standard and comparison stimulus in succession, and was asked if the comparison stimulus was greater than, equal to, or less than the standard stimulus. The parameter was varied by equal increments in the comparison stimuli. Gescheider (1976) describes the determination of the difference threshold for loudness for a 20 db 1,000 Hz tone. The comparison stimulus was varied 5 db above and below the standard in .5 db increments. Some trials were started from the lowest (15 db) level, and were alternated with trials starting from the highest level. The upper limen was the point on the physical dimension where the "greater" responses changed to "equal" responses. The lower threshold was the point where the "lesser" responses changed to "equal" responses. The point of subjective equality was half of the mean upper threshold plus the mean lower threshold. The interval of uncertainty was the area on the stimulus dimension over which an observer could not perceive a difference between the comparison and standard stimuli. Half of this interval was considered the difference threshold for that stimulus. Usually six to eight trials were averaged for each standard stimulus level.

The method of limits is subject to the effects of errors of habituation and expectation, and to errors due to time (Warren, 1970; Gescheider, 1976). To attempt to counterbalance for these effects, the standard and comparison stimulus are usually placed in a random order of presentation.

The method of limits in its classical form cannot be used to measure the difference threshold of timbre for the same reasons that the method of constant stimuli is unsuitable. However, since the concern

in the method of limits is with transitional points as related to a stimulus value, rather than with the direction of the stimulus magnitude, modifications of the procedure have promise for investigating differential sensitivity for stimuli which do not lie on a sensory continuum.

Research with Multicomponent Stimuli Critical Bandwidth

Much of the research in contemporary psychoacoustics has utilized sinusoidal tones. Recently, research with multicomponent tones has increased. Much of this research has direct bearing on this study.

One of the most significant studies with multicomponent stimuli was carried out by Zwicker (1954). He investigated the masking effect of two simple tones f_1 and f_2 on a narrow band of noise with a center frequency of $\frac{1}{2}(f_1 + f_2)$. The two simple tones were of 50 db SPL at equal frequency distances above and below 570 Hz. The probe sound was a band of noise with a bandwidth of 30 Hz centered on 570 Hz. One subject was in the experiment. The masked threshold was measured with the von Bekesy up-and-down tracking technique used in a modified method of limits format. The subject was required to press the button on an automatic attenuator if the probe sound was audible and to release the button if the sound became inaudible. The position of the attenuator was continuously recorded on a strip chart mechanism. This record was used to determine the masked threshold.

Zwicker found that the masked threshold was constant for frequency separations less than 130 Hz, but that the threshold decreased

progressively beyond that value. The masked threshold at $f_2 - f_1 = 50$ Hz was 35 db. This value was maintained until about 130 Hz. At $f_2 - f_1 = 200$ Hz, for example, the masked threshold was found to be 33 db. Zwicker termed the frequency difference at which the change in masked threshold occurred the "critical frequency difference," or "critical band." The procedure was repeated with two subjects for other frequencies. A graph was made plotting critical bandwidth against the center frequency of the masker. This allowed one to perceive the size of the critical bandwidth as a function of frequency. This experiment thus provided an estimate of the ear's analytical power. The ear, with respect to this study, is viewed as a set of adjacent filters of a given bandwidth, that bandwidth being the critical band.

Gassler (1954) determined the difference threshold for a probe tone masked by another tone. The pure tone threshold for a standard was found using the Von Bekesy method of limits procedure. Another tone of equal intensity was added to the standard, but had a frequency of 10 Hz lower than the first tone. The threshold was again found using the Von Bekesy tracking procedure. Additional tones were added until up to forty components of various spacings and various frequency regions were used. Two subjects participated in the experiment. Gassler found that, up to a certain point, the amplitude of the individual tones decreased as more and more of them were added, but after a point no further decreased occurred. The point at which this happened was found to approximate the critical bandwidths found by Zwicker (1954).

Greenwood (1961a, 1961b), using subjects who had at least two months of training, conducted experiments to confirm Zwicker's

results. He used pure tones monitored with an electronic counter. Signals were interrupted by an electronic switch for presentation at three pulses per second. Each signal had a rise-fall time of 25 msec. Noise was ring modulated with a carrier to obtain two sidebands centered around the carrier frequency. A wave analyzer was used to monitor the resulting waveform. The Von Bekesy technique was employed to determine the difference threshold. Each session lasted two hours with a ten minute break between sessions.

Greenwood found Zwicker's estimates of the critical band at frequencies greater than 1000 Hz to be larger than his results, which indicated a correspondingly smaller critical bandwidth estimate above this frequency.

Green (1965) found larger critical bandwidth estimates than either Zwicker (1954) or Greenwood (1961a, 1961b). The maskers were two sinusoidal tones of 77 db SPL. The signal was gated for a presentation time of 124 msec with a linear rise-fall of 12 msec. A two-alternative-forced-choice (2AFC) procedure was used. The signal occurred at random in either of two temporal intervals which were marked for the observer by lights. For each condition of the experiment, five signal levels separated from each other by two db were used. At each of three center frequencies (250 Hz, 1000 Hz, and 4000 Hz), the frequency separation of the maskers was varied from a minimum width of four Hz to a separation exceeding the frequency response characteristic of the headphones. Observers were told which of their responses were accurate. They were trained extensively for two weeks and paid on an hourly basis.

A psychometric function was plotted relating the percentage of correct decisions to the signal level. The .75 level was used as a criterion. The results of the experiment indicated a masked threshold for a single masker at 80 db SPL was 80-75 db at 250 Hz, 65-70 db at 1000 Hz, and 70 db at 4000 Hz. These results indicate a critical bandwidth two times smaller than that reported by Zwicker (1954), as the amplitude levels required for masking are twice as high.

Zwicker and Fastl (1972) repeated experiments by Green (1969), Elliott (1976), and Scholl (1962) which determined that the critical bandwidth varied with time after the onset of the masker. Two subjects listened for a probe tone with a gated masker and continuous masker. The probe tone was 1,000 Hz signal. The maskers were at 70 db SPL, and ranged in frequency from 300 to 5,000 Hz. Zwicker and Fastl found that differences between the difference thresholds for gated and continuous thresholds disappeared if a 1/3 octave band filter was used to narrow the energy spread of the gated signal.

These studies indicate that the resolving power of the ear is no better than about 1/3 octave. The experiment of Zwicker (1954) is badly in need of replication with a much larger sample size. Until this is done, it will not be possible to speculate whether the results of Green (1965) with a single masker are more acceptable. Evidence from experiments in lateral suppresion characteristics of the ear suggest that they are. However, experiments in loudness calculations for complex sounds are not in total agreement with the lower estimates of Green (1956). These topics will be dealt with later in this chapter.

A study using multicomponent stimuli in which each of the components is varied, such as the present study, must take into account the limited resolving power of the ear. The implication from the studies of Zwicker, et al is that variations in the intensity of two partials of an inharmonic complex might not be discriminated if these partials lie within the critical band. Therefore, harmonic complexes with relatively low partial numbers, each lying in a separate critical band, might be more suitable for the present study.

The interpretation of the ear as a perfect linear filter is questioned by the results of some recent experiments. Von Bekesy (1963) asked subjects to match a probe tone with the upper and lower cut-off frequencies of an octave band of noise routed through a bandpass filter. The filter had a bandpass of 400-800 Hz. Subjects made ten consecutive matches each. The average adjustment of the probe tone was 398-399 Hz for the lower cut-off frequency and 804-805 Hz for the upper cut-off frequency. Von Bekesy considered this as demonstrating that the edges of a band of noise are emphasized by contrast phenomena similar to the Mach bands in vision. These results are supported by Carterette, Friedman, and Lowell (1970), who demonstrated the masked threshold of a pure tone as a function of frequency in the presence of a band of masking noise which had very sharp edges at the lower and upper cut-off frequencies. Their results indicated a slight increase in the masked threshold near the edges of the noise spectrum; they interpret this as indicating the existence of Mach bands in hearing, due to lateral inhibition.

Houtgast (1971, 1972, 1973) conducted a series of investigations which describe the nature of lateral suppression in the ear. Houtgast argues that nonsimultaneous masking techniques must be applied in studying lateral suppression, because if both the masking noise and the probe tone are subjected to lateral suppression effects, the masking countour will not demonstrate these effects.

Houtgast (1971) used two subjects in an experiment designed to contrast direct with gap masking techniques. The three masking stimuli were bands of noise at 60 db SPL. The masker had a variable cut-off frequency. The probe frequency was at 1500 Hz. The masker was presented in a rhythm of 150 msec on and 50 msec off. The test tone bursts, each 20 msec long, were presented in the gaps. The Von Bekesy tracking technique was used. When the level of the test tone was just above the threshold, the series of tone bursts sounded continuously. This was termed the pulsation threshold. The median level during fifty presentations defined the threshold value.

Although no statistical evidence was offered, the curves for direct masking and gap masking appeared different. The direct masking curves were smooth and thus suggest the operation of a linear filter. The gap masking curves, however, had sharp edges, suggesting a non-linear filter of the signal-compressing type.

Houtgast (1973) supports the evidence for lateral suppression by suggesting that a tone should be able to suppress to a certain extent another simultaneous tone of close frequency. Five different experimental paradigms were used with two subjects. The stimulus consisted of a weak tone of 1000 Hz at 40 db SL. A stronger suppressing tone ranged from

1000 to 2500 Hz, and was at 60 db SL. Houtgast found that for direct masking paradigms the detectability as the suppressing tone approached 1000 Hz increased. Non-simultaneous paradigms revealed that the detectability decreased abruptly to a minimum at 1150 Hz. Thus a tone of 40 db SL at 1150 Hz can reduce a 1000 Hz 40 db SL tone by 20 db in threshold. Houtgast (1973) also found that tone higher in frequency suppresses a lower tone substantially more strongly than conversely.

The evidence for lateral suppression in the ear illustrates the complexity of the filter characteristic of the ear. Direct masking techniques show excellent agreement in results for the masked threshold of noise (Zwicker, 1954; Houtgast, 1971). The bandwidth of the auditory filter when determined by non-simultaneous masking reveals values about half of what might be expected. The intrusion of lateral suppression into an otherwise linear system casts uncertainty into any experiment dependent upon critical bandwidth calculations. Any use of direct-masking ciritical bandwidth estimates should be toward a conservative bandwidth.

Loudness

A given sound complex can yield sensations of timbre, pitch, and loudness. Investigations into the parameters which control these sensations have been confined largely to the last 25 years. The results indicate a complex, if not confusing, relationship between three physical parameters and the elicited sensation. The power spectrum is such a parameter; it effects loudness, timbre, and pitch. Important to this study is an understanding of the loudness sensation produced by a complex sound.

Zwicker, Flottorp, and Stevens (1957) investigated the relation of power spectrum to the loudness sensation. The complex stimulus was generated by four oscillators. The four-component complexes were centered on 500, 1000, and 2,000 Hz. The tones were presented binaurally through headphones at 57.5 db SPL. The subject adjusted the level of a sinusoidal comparison signal to match the loudness of the tone complex. The two sounds were presented alternately for about one second each with .5 second silent intervals between them. The reverse procedure, in which the complex stimulus was matched to the sinusoidal stimulus, also was used. The level of the complex stimulus was calculated by measuring a single component. In all cases the center frequency of the tone complex was equal to the frequency of the comparison sinusoid. The independent variable was the spacing of the four tones of the complex stimulus.

Results showed that when the overall spacing of the compenents of the complex stimulus reached a critical value, the subjective loudness increased. Superposition of the graphs of masking experiments (Zwicker, 1954) with these data showed that the data were predicted from the critical bandwidth estimates.

Graphs of loudness vs. spacing and level showed that the estimated critical bandwidth was independent of level. However, for a complex sound of 17.5 db SPL the results suggested that the critical band may make loudness decrease at low levels as spacing increases within the band.

Experiments with bands of noise showed that noise bands with a center frequency of 1420 Hz and a bandwidth from 20 to 10,000 Hz do not

evidence increased loudness when the bandwidth of such noises exceeds the critical bandwidth.

Zwicker, Flottorp, and Stevens (1957) further state that when two sounds differ in quality, the matching of their loudnesses was difficult and subject to considerable variability. They also suggested that the difference threshold for frequency of sinusoidal tones is a constant fraction of the critical band, indicating a correspondence to equal distances along the basilar membrane. The model presented thus is one of a set of adjacent auditory filters.

Scharf (1959) investigated the loudness of complex sounds near threshold. One-hundred subjects, eight to twelve in each experiment, were used. The stimuli consisted of complex sounds with four components of equal loudness level. This LL was determined separately for each subject. The fundamental frequency was the independent variable, and was centered on 500, 1,000, 2,000, and 3,000 Hz. The complexes were matched with sinusoidal tones by a forward-masking procedure. The signals were alternated on and off at one second intervals. Comparison sound threshold was measured by a modified method of limits. Comparison stimuli were presented at a sound level of 15, 25, and 35 db. The complexes were adjusted to equal the loudness of the sinusoid. Scharf found that spreading of energy over more than a critical band increased the loudness at sensation levels above 10 to 15 db, but that loudness was unchanged or decreased below that level.

These results indicate that confounding factors surface in the analytical power of the ear at low intensity levels. The reason for this

is not clear, however Plomp (1976) explains:

> If a level of difference of 10 db corresponds to a factor of two in loudness . . . we may expect the asymptotic plateau for four widely spaced tones to be 20 - 6 = 14 db above the level for close spacing . . . . This reasoning can also clarify why the effect of frequency separation is zero or even negative at low sensation levels (p. 79).

It is clear that moderate intensity levels will insure that the analytical power of the ear corresponds most accurately to results obtained during direct-masking and loudness summation experiements, and that the sound pressure level chosen in timbre discrimination investigations should be greater than 20 db, and optimally 50 to 70 db.

Scharf (1959) also investigated the loudness of complex sounds as a function of the number of components. Scharf chose complex sounds comprised of 2, 3, 4, and 8 intense tones which were matched in loudness to a 1500 Hz tone at 25, 50, 75, and 90 db SPL. Frequency spacing between components was equal. The tones were produced by mixing the output of oscillators. SPL was calculated from the voltage level measured across headphones. The tone complexes were centered at 1500 Hz. Spacing between components was 1600 Hz (greater than the critical band) and 176 Hz (less than the critical band). Complex and comparison signals were presented alternately at one second intervals. The stimuli had rise and fall times of 35 msec to avoid transients. The subject controlled the intensity of the complex or comparison signal by adjusting a potentiometer. Each subject matched the complex stimulus to the comparison stimulus and vice versa two times.

Results indicated that the loudness did not depend on the number of components. Although mutual inhibition increases when the frequency

spacing among components is decreased by the insertion of more tones into a complex, a concomitant increase in the number of critical bands that contribute to the total loudness may offset this inhibitory effect.

Scharf (1963) also found that loudness summation was dependent on spectrum shape. In this investigation, the stimuli were three-component complexes with a center frequency of 2,000 Hz. The loudness level was determined by the method of adjustment using a 2,000 Hz comparison tone. Six subjects participated. The intensity relations among the components were varied in three paradigms. In one paradigm, the components had a positive slope, in one a negative slope, and in the third a peaked condition was generated. Spacing of the components was 280, 1,100, or 2,200 Hz. Either the sensation level of the comparison stimulus or the sound pressure level of the complex sound was held constant.

Results indicated that in those three-tone complexes whose overall component spacing was greater than a critical band, the reduction of the intensity of one or both side tones caused the loudness of the tone complex to decrease. Supercritical complexes were loudest when components were equally intense, and were at a moderate level of 60 db SPL. Masking effects were evidenced in the sloped spectra. Results indicated the -25 db slope at 80 db SPL was louder than the +25 db slope. This is due to the fact that masking effects are greater at higher frequencies.

These results indicate that the adjacent critical bands do not contribute equally to the sensation of loudness. The ear is not a perfect bandpass filter. Whether this indicates that the power spectrum of a complex sound yields a sensation of timbre in which the components

provide unequal contributions is not clear; this may be clarified by the present study. It is evident that the alteration in the intensity of a single component of a complex sound can give rise both to a change of loudness and a change in timbre. Whether one can separate the loudness and timbre sensations with reference to power spectrum remains to be seen.

Several methods have been devised for quantifying the magnitude of the loudness sensation of a complex sound. Zwicker and Scharf (1965) determined a model of loudness summation based upon psychophysical measures of the auditory process. In essence, the procedure involves determining the loudness level contribution of each of the critical bands. Twenty-four contiguous critical bands span the audible range from low through high frequencies. The excitation pattern of the stimulus is first expressed in db as a function of the logarithm of the frequency, and is determined to be 3 db above the stimulus's masking pattern in db SPL for lower and middle frequencies and 6 db above the masking pattern for higher frequencies. A frequency-dependent constant is used to correct for the transmisson characteristic of the middle ear above 2,000 Hz. The frequency scale of the masking pattern is converted to a scale based on the critical band. Then for every critical band the excitation pattern is related to its contribution to the loudness of the stimulus. This is done in relation to a power function derived from techniques established by Stevens (1961). To determine the loudness of the complex sound in sones, the area under the specific-loudness curve is plotted as a function of the critical band frequency scale and is integrated over a series of values.

Following the empirical approach of Stevens (1961), ANSI has adopted third, half, or full-octave bands, as a base for loudness computation of complex sounds. In this procedure, the sound is measured in bands. The sound pressure level of each band is entered into a calculation nomograph which determines the loudness index for each band. Loudness in sones is equal to the largest loudness index plus a constant times the difference between the total sum of loudness indices and the largest loudness index (Equation 7).

Stevens (1972) updated this procedure by referencing 25 experimental contours. Using as a reference a 1/3 octave band of noise centered at 3,150 Hz, Stevens found that perceived magnitude of loudness grew as a 2/3 power of the sound pressure, doubling for each 9 db increase. The summation formula remained the same as in the 1961 procedure (Equation 7), but F was made to vary as a function of level to reflect the nonlinear growth (log-log coordinates) of broadband noise. Results showed that perceived level in db (PL db) was about 8 db lower than the older loudness level in phons calculated using the earlier procedure.

The second procedure ignores several important considerations. It does not take into account the spreading of energy to different critical bands, the masking phenomena reported by Scharf (1959). The calculation formula also simply treats the total energy in a band, ignoring the distribtion within the band. As long as the octave band is as small as a critical band, this assumption has been experimentally verified (Scharf, 1959b; Zwicker, Flottrop and Stevens, 1957). The second procedure, however, does not require the use of computer programs.

These results indicate a means of computing the loudness level in sones for complex tones of varying sound spectra, and are useful in relation to timbre studies which also are based on sound spectra alterations. It appears that the extent to which loudness will confound timbre perception as related to power spectrum will be dependent on the specific parameters of the complex sound.

In summary, the ear is seen in loudness calculations as an auditory filter of a given bandwidth. When components of a complex stimulus are within a critical band, the loudness is related to the total energy flow in that band. If the components of a complex sound exceed the critical band, the subjective loudness is greater than that predicted by the above equation, tending toward a value given by the sum of the individual loudnesses related to the energy within each band. Nonlinear distortion and lateral suppression effects appear at low intensity levels, and masking effects appear when individual loudnesses between bands vary considerably.

Both timbre, as defined by Equation 1, and loudness, as defined by the above studies, are dependent on power spectra. It would appear, therefore, that these two characteristics are interdependent in complex sounds. A study determining the precise relationship is called for, as the writer has yet to find any literature relating to it. In the present study, in which the power spectrum of a complex sound will be systematically altered, loudness effects easily could influence results. This is a negative finding only if one considers loudness and timbre as two distinct characteristics. By the definitions in this study, the two areas are not distinct. The extent of this blur should be investigated.

Resolution of the Individual Components

The ability of the ear to resolve (hear out) the individual components of a complex sound has been investigated since the time of Ohm (1843) and von Helmholtz (1877). Helmholtz (1877) was aware of the difficulty in hearing the components of a complex tone, and suggested that a probe tone of the same frequency as the component be sounded to aid the subject in identifying the tone. This important consideration has been used in the last century with much success.

Thurlow and Bernstein (1957) presented pairs of simultaneous sinusoids to subjects. The frequency differences between the components covered a wide range. The subjects were asked whether one or two pitches were audible. Results indicated that at low frequencies in particular, the individual tones of a pair can be identified for substantially smaller frequency separations than the partials of a complex tone.

Plomp (1964) conducted a series of investigations to expand upon and verify Thurlow and Bernstein's results. In the first experiment, two component stimuli with various frequency separations were presented to two subjects in a forced-choice procedure. The frequency of one probe tone coincided with one component of the stimulus, whereas the frequency of the other probe tone was either 1/4 Δf below the higher or 1/4 Δf above the lower component of the stimulus, with Δf = frequency difference between the components. At 200 Hz, only about 20 Hz frequency separation was needed to match the appropriate probe tone. The fundamental frequency was varied from 44 to 2,000 Hz. Results confirmed those of Thurlow and Bernstein (1957).

The second investigation involved a complex stimulus consisting of twelve cosine components with phase angles in multiples of 6°.

Each component had a loudness level of 60 phons and was generated by a function generator. The output signal was checked with a wave analyzer which indicated that amplitude differences between the partials did not exceed .5 db SPL. The function generator output was passed through a lowpass filter to eliminate all harmonics above the twelfth. Signals were presented monaurally. A forced-choice procedure was used; after each decision, the observer was told whether his response was correct or not. All values were tested in random order. The subject could select three stimuli by means of a three position switch. The middle position presented the complex stimulus, and the outer positions presented probe tones, one with frequency nf_0 (n = integer), the other one with frequency $n \pm f_0$. The subject was allowed to switch freely from one position to another, and was required to judge which of the two probe tones also was present in the complex sound.

The results of six intensively trained subjects showed that observers could distinguish the first 5 to 8 harmonics, with slight dependence on fundamental frequency. The most distinguishable harmonics were for a fundamental frequency of 126 Hz. The experiment rejected the view that odd partials were easier to hear, as suggested by Helmholtz (1877). Plomp suggests that the slope of the excitation pattern that is not masked by a neighboring component contributes to the audibility of the partial.

Plomp and Mimpen (1968) repeated Plomp's (1964) investigation, and also investigated the identifiability of the partials in inharmonic complexes. An audiometer was added to the equipment of Plomp (1964) so that the harmonics of the complex tone could be presented all at the same sensation level. Stimuli were presented to subjects at 60 db SL.

In the first replicative investigation, for each fundamental frequency between 44 and 3,000 Hz the percentage of correct responses diminished monotonically for the increasing harmonic number. The percentage of correct responses was plotted as a function of harmonic number and was fitted by a smooth curve of which the 75 percent point was taken as the limit of the number of distinguishable harmonics. The results of six subjects indicated that for a fundamental frequency of 500 Hz the first five harmonics could be identified. Thus the improved procedure yielded slightly more conservative results than those obtained by Plomp (1964).

Eight subjects were used to investigate inharmonic complexes. The complex stimuli were two series of 12 inharmonic partials distributed evenly between 230 and 2,200 Hz. The same procedure as in the first investigation was used. Results were statistically similar to those obtained with harmonic complexes. Data points indicated that there were substantial differences among subjects in their ability to identify partials. The data also confirm the critical band as a measure of the analytical power of the ear, at least in middle to high frequency ranges. The data of the second investigation of Plomp and Mimpen (1968) showed that for mid- to high-frequencies the separation between the partials must be a constant percentage of the frequency of about 15 to 20 percent. Data plotted as frequency difference between harmonics of a complex tone just required to hear them separately, as a function of frequency, produced curves which approximate those obtained by Zwicker (1952).

Pollack (1964) used a similar procedure. Subjects were requested to decide whether a probe tone, presented after a complex

stimulus, was or was not a member of the stimulus. Subjects were paid university students with previous experimental backgrounds. Eight different paradigms using one, two, and three tone complexes and simultaneous and nonsimultaneous probe tone presentations were employed. Eight hundred trials were given over a two hour period interspersed with rest. Tones were presented for 100 msec.

The results of the 2AFC procedure indicated correct responses to 52 percent of the four-tone complexes in the simultaneous paradigms and 73 percent correct for nonsimultaneous paradigms. The results were so poor that the author questioned the validity of Ohm's acoustical law.

However, the experimental procedure was inferior to Plomp's (1964). As Helmholtz (1877) suggested, the observer's attention should be drawn to the harmonic to be identified. The 2AFC procedure does not take into account this requirement.

These results suggest that in any investigation using complex sound stimuli the attention of the observer may be directed at a given component of the sound. In timbre investigations, it therefore seems important not to allow the attention of the individual to focus on one of the first five to eight harmonics, since the perception of a fused whole is desired. By using a 2AFC procedure, it appears that such attention will be minimal when compared with the method of adjustment types of procedures.

Several investigators report identification of partials beyond those indicated by the above authors. When analyzed the differences are due to procedural considerations, and have important implications in confirming the role of lateral suppression in the hearing process.

Schouten (1940) used a procedure which eliminated and reintroduced a given partial repeatedly. With low fundamental frequencies, partials up to the 16th could be heard. This procedure does not indicate the resolving power of the ear. It merely demonstrates the ability of the ear to respond to differences in the amount of stimulation, and does not indicate whether the separate components of the steady-state pattern are being resolved.

In a similar procedure, Gibson (1971) generated complex sounds with phase-locked oscillators. Components of the sound complex were adjusted for uniform sound levels. The resultant complex tone could be deprived of any partial by means of an electronic switch. The observer was asked to duplicate the frequency of the interrupted partial. The chosen frequency was checked with a frequency counter. In middle frequencies results indicated that the 11th and 12th partial could be heard. This investigation suffers from the same methodological concerns as Schouten's (1940).

Duifhuis (1970) used periodic pulse trains to investigate the extent of partial identifiability. Complex sounds were created by generating a sinusoid of f Hz, equal in frequency to the highest harmonic of the periodic pulse. An adjustable divider divided the frequency f_1 of the sinusoid by an integral factor and triggered a pulse shaper. The latter yielded a periodic pulse with a rate of g Hz ($f = ng$) and a width of 100 msec. The pulse rates were at g = 25, 50, and 100 Hz. Harmonic numbers investigated were n = 5 to 60. Thresholds were measured in db SPL. The pulse train had a SL of 30 db. Measured threshold values were normalized so that A = 1 indicated the amplitude

of the unaffected harmonics, The subject had to discern one particular component, characterized by a given pitch, from the complex sound of periodic pulses. The threshold was the amplitude value at which the harmonic involved was just audible as a pure tone. The subject varied the amplitude of one individual harmonic by manipulating attenuators.

Graphs of normalized threshold amplitude vs. frequency of harmonic indicated three regions of identifiability. The first was for n less than 10, the second for n between 10 and 28, and the third for n greater than 28 harmonics. Duifhuis reported that for lower harmonic numbers (region 1 and 2) the ear demonstrates frequency analysis as expected. However, for region 3 cases, time analysis became important. This was due to the limited frequency resolving power of the ear, as found by Houtgast (1973) in his studies on lateral suppression using the pulse-threshold method of determining the critical bandwidth. When the pulse trains were at rates less than the critical bandwidth, there was a separate response to the successive pulses. The eliminated harmonic was actually added in the silent intervals, and became audible as a pure tone.

Again, the importance of simultaneous vs. nonsimultaneous approach to investigations dealing with the resolving power of the ear is evident. Lateral suppression and figure-ground effects play a major role in results from experiments using components varied against a continuous background or components altered within a train of pulses. A 2AFC procedure using comparison and standard stimuli seems to avoid the considerations arising from the limited frequency power of the ear.

Combination Tones

Since a complex tone is in reality a series of simultaneous sinusoids, it is important to consider the impact of combination tones in research dealing with complex tones. The results of experimentation in this area provide a fundamental understanding necessary to comprehend the effects of phase on timbre.

A purely linear model of the basilar membrane does not consider the possibility of combination tones. An example of a combination tone is one of the class $f_2 - f_1$, commonly called the difference tone. If $f_1 = 100$ Hz and $f_2 = 150$ Hz, a combination tone of 50 Hz is heard. Although attempts have been made to suggest that these tones have a physical nature, recent research indicates that the origin of such tones is within the ear itself. Helmholtz (1877) was the first to suggest the possibility that combination tones are produced by nonlinear effects in the middle ear.

Plomp (1976, p. 27) gives a detailed description of the creation of combination tones by a nonlinear transfer characteristic. In essence, the displacement on the basilar membrane d is given by the relation $d = a_1 p + a_2 p + \ldots a_n p$, rather than by the linear relation $d = ap$, (where a_n is amplitude and p is pressure of the components). In general, the term p^k introduces harmonics and combination tones with frequencies $mf_2 \pm nf_1$ and amplitudes proportional to $p_1^n p_2^m$ with $m + n = k$.

Zwicker (1955) investigated the above relations via combination tones of the class $f_2 - f_1$ and $2f_1 - f_2$. Cancellation procedures were used. A simultaneous probe tone with the same frequency as the combination tone was introduced. The subjects were required to adjust the amplitude and phase of the probe tone so that the combination tone

vanished. The frequency of f_1 was 1,000 Hz; f_2 was 1400 Hz, therefore the combination tone was at $f_2 - f_1 = 400$ Hz. For an intensity of 100 db SPL for the second component of the pair, the cancellation level of the 400 Hz probe tone was about 40 db below the level of the first component. Cancellation was independent of the frequency separation of the primary tones, f_1 being constant.

The results with this class of combination tones were completely as predicted. The amplitude of the combination tone was proportional to $p_1 p_2$. This relation predicts that the level of the combination tone should increase 10 db for every 10 db increase in either component, and 20 db for a 10 db increase in both components.

However, Zwicker's (1955) results with combination tones of the class $2f_1 - f_2$ were in complete disagreement with the expected results. The amplitude of the combination tone should have been proportional to $p_1^2 p_2$, implying that its level would increase 20 db for a 10 db increase of the first component, 10 db for a 10 db increase of the second component, and 30 db for a 10 db increase in both components. Results plotted by Zwicker (1955) showed much smaller values which were dependent upon the frequency separations of the primary tones. The cancellation level, indicative of the greater amplitude of the combination tone, was higher for small frequency separations than for large frequency separations.

Plomp (1965) investigated the detectability threshold (absolute threshold) of combination tones. Subjects were presented with a two-tone stimulus which was alternated with a probe tone of variable

frequency. The subject was required to search for all combination tones with a pitch equal to the pitch of the probe tone. The primary tones were at f_1 = 1,000 Hz and f_2 which varied between 1,000 Hz and 2,000 Hz. The two tones were reproduced by separate loudspeakers to avoid intermodulation products in the stimulus. Four levels of intensity were used: 70, 60, 50, and 40 db SPL.

The results of four subjects indicated that only combination tones below 1,000 Hz were heard except by one of the subjects. More combination tones were heard for high SPL than for smaller intensities. The only combination tones heard by all subjects were $f_2 - f_1$, $2f_1 - f_2$, and $3f_1 - 2f_2$. More combination tones were heard for small frequency differences than for larger ones. Subsequent investigations revealed that there were large inter-individual differences in the minimal sensation level of primary tones at which the combination tones became audible. Levels of 50 to 60 db SPL were required to hear the difference tone, which was the easiest to hear due to its lower f_1 frequency.

Smoorenburg (1972) also investigated the audibility of combination tones. He used sine generators to create the primary components. The lower frequency stimulus component was fixed at f_1, the higher at $2f_1$. The method of adjustment was used; the subject turned the dial of an oscillator down until the combination tone appeared.

Results indicated that at a low SPL of 40 db all subjects could hear $2f_1 - f_2$ over a restricted frequency range of about 800 Hz. For three subjects, independent of f_1, the combination tone $2f_1 - f_2$ became audible at sensation levels of the primary tones as low as 15 to 20 db. Higher order combination tones of the class $f_1 - n(f_2 - f_1)$

became more audible as f_2 approached f_1 in frequency. The lower frequency limit of combination tone audibility was approximately independent of n. At 40 db SPL for $f_1 - f_2$ the limit was about $.6f_1$ for low and high values of f_1. Combination tones up to $n = 6$ occasionally could be observed.

Smoorenburg (1972) also used forward-masking and pulsation threshold techniques for determining the level of combination tones. The results indicated that for $f_2/f_1 = 1.3$ the cancellation level was significantly higher than for estimates based on nonsimultaneous procedures. Smoorenburg (1974) reported the results of one subject in cancellation vs. threshold paradigms for $f_1 = 1{,}000$ Hz and $f_2 = 1{,}122$ Hz in which the cancellation procedure gave a 20 db SPL higher threshold estimate. He suggests that rather than speaking of a polynomial transfer function such as Plomp's (1976, p. 27), model, a modification is required in the amplitude terms. Rather than having a constant multiplied by pressure, his investigations indicate that the constant is dependent on stimulus amplitude, and that an adaptation mechanism is at work. Smoorenburg (1974) further suggests that the ear is a filter that compresses signals in such a way that cancellation levels for $2f_1 - f_2$ increase proportionally to the level of the second component if this level is less than the level of the first component and decreases for a level of the second component greater than the first component. A complete mathematical description is given in Smoorenburg (1974).

The results of these experiments with combination tones indicate the complexity with which the ear functions in translating the vibrations of complex stimuli. Even with simple cases, such as

combination tones resulting from two components, the ear's nonlinear aspects introduce complexities in the relation between the physical and the psychological. It is hard to imagine the number of combination tones generated by a tone complex of 16 inharmonic components, let alone the masking effects and beating effects of these combination tones. Prediction of the amplitude of combination tones of the higher classes is difficult at the present stage of research and must await further empirical evidence upon which to base a new mathematical function. Combination tones of the class $f_2 - f_1$ may have interesting consequences for harmonic tone complexes where the components are defined by nf_0 (integral multiples of the fundamental). If the fundamental is eliminated, the third harmonic and second harmonic produce a combination tone at the fundamental frequency. The amplitude, of course, of this apparent missing fundamental will be proportional to the amplitude of the second and third harmonics.

At this time, it is difficult to comprehend the possible relation of the various combination tones to the primary components in a complex sound's ability to elicit the sensation of timbre. It appears to the writer that this is another dimension that can be considered by future investigators in the quest for understanding the timbre sensation.

Low Pitch

Seeback (1841) was one of the first experimenters to investigate the phenomenon of low pitch. Using mechanical sirens to produce different waveforms, he thought that the physical correlate of the pitch was the rate of air puffs through the holes of his metal discs. Three

metal discs were constructed: The first disc had a single hole near the edge, the second disc had a second hole immediately opposite the first along the diameter, and the third disc had a second hole opposite the first, but to the right of the diameter. The discs were rotated and air was passed through the holes. Disc 1 had a frequency equal to 1 / t where t = time/sec. Disc 2 had a frequency equal to 2 / t. Disc 3 produced a complex tone with components 1 / t and 2 / t. However, the amplitude of the first component was very low. Yet, subjective evaluation of the pitch produced by the discs indicated that the third disc produced the same pitch sensation as the first disc, even though this frequency component was not present.

Schouten (1938) began to systematically investigate this phenomenon. Also using sirens, he generated a complex sound with a fundamental frequency of 200 Hz in which the fundamental was canceled. A probe tone of 206 Hz was sounded and caused no beats, yet the pitch of the complex was the same as a complex with the fundamental. Schouten (1940) conceived that the lower harmonics were sounding separately, and that the unresolved higher harmonics created what he termed the "residue pitch," which coincided with the pitch of the fundamental.

Ritsma (1962) investigated the existence region of the tonal residue. He used complex stimuli of three harmonics created by amplitude modulation of a carrier of nf Hz with a simple tone of f Hz. The sounds were presented to a subject at 35 db SL. For n = 4, 5, 6, . . . Ritsma determined the minimal modulation depth required for hearing a low pitch. He then plotted the maximal center frequency nf against

fundamental frequency, and drew curves representing various modulation depths,

Results indicated that a decrease in modulation depth shrank the existence region of the sensation of low pitch. He found that low pitch required that at least three harmonics were situated within the existence region for a three component complex. The lower harmonics, below about $n = 7$, gave a much better low-pitch sensation than the higher harmonics. This conclusion was supported by a later investigation (Ritsma, 1967).

Plomp (1967) determined the main components of low pitch by a different procedure requiring two complex stimuli with twelve cosine terms. Stimulus 1 was a standard; stimulus 2 was identical, except that the fundamental frequency was lowered by 10% in frequency and the other partials were increased by 10% in frequency. The fundamental was varied from 125 Hz to 2,000 Hz. The stimuli were presented monaurally by headphone in random order at 60 db SPL in 200 msec bursts. The subject had to indicate which stimulus had the higher pitch.

The results of 14 subjects indicated that up to about 1,500 Hz, the second and higher harmonics determine the pitch sensation. By plotting frequency of the fundamental vs. harmonic number from Ritsma's (1967) data, Plomp showed that for a fundamental of 500 Hz, the third harmonic is dominant for determining low pitch. Thus for a given fundamental frequency, various dominant regions are mainly responsible for creating the low pitch sensation.

Smoorenburg (1970) investigated the role of combination tones in low pitch perception. His stimuli consisted of an inharmonic complex

sound of two components. The subject was required to match this in pitch with a harmonic two-component stimulus. The stimuli were presented monaurally by headphone at a level for the individual components of 40 db SL.

The results indicated that combination tones, introduced by the ear, contributed to the low pitch sensation. This was true of both harmonic and inharmonic tone complexes.

Houtsma and Goldstein (1972) conducted investigations into the central pitch processing of complex tones. Stimuli were two randomly chosen successive harmonics with equal SPLs presented both monaurally (two harmonics to each ear) and dichotically, (one harmonic to each ear). No energy was provided at the fundamental frequency. The task of the subject was to identify on each trial which of 8 known two-note melodies were presented. The melodies had identical envelopes and time structures and were tuned to the natural scale. They were presented at 20 db SL.

Results showed equal accuracy in both the monotonic and dichotic presentations. Results imply that the pitch of complex tones is mediated by a central processor operating on neural signals derived from those stimulus harmonics that are tonotopically resolved.

Whether low pitch is derived from spatial or temporal information during processing by the ear remains an unresolved question. Goldstein (1973), Terhardt (1974), and Wightman (1973) present three different models which are summarized in detail by Roederer (1975, pp. 174-182).

It is clear that several different aspects of a complex sound contribute to low pitch. If the fundamental is missing, the lower resolved harmonics play an important role. If these harmonics are missing, combination tones may produce the low pitch effect. The fundamental itself is not as important in the sensation of low pitch as the other components. Thus in an experiment in timbre that emphasizes or eliminates partials from the overtone structure, one would expect that, due to various mechanisms, the low pitch would remain the same and not confound the results.

Studies in Timbre

Early Work

Seebeck (1849) was one of the first to suggest that timbre originates from the harmonics of a complex tone, but it was Von Helmholtz (1877) who conducted extensive experimental research into timbre generation and perception.

Von Helmholtz (1877) used an appratus consisting of tuning forks excited by intermittent electrical currents conducted through the coils of an electromagnet. The amplitude of the waves emitted by the tongs of the fork was controlled by the use of resonance chambers fastened on a track near the tuning fork.

He noticed that the simple vibrations produced by the forks seemed to have their own characteristic timbre. Simple tones sounded sweet and pleasant; dull at low frequencies and brighter and sharper at high frequencies. For the study of complex sounds, he used eight tuning forks. The fundamental frequency was 120 Hz, and the upper

harmonics were integral multiples of the fundamental.

Von Helmholtz found that most vowel-like sounds could be produced by complex tones with more relative amplitude in the lower harmonics. He found such tones were more "musical" and rich than tones with high amplitude upper harmonics. Tones with only odd harmonics produced the "nasality of the clarinet," and tones with all harmonics at equal amplitudes produced "the softer tones of the horn." Tones with relatively high amplitudes in the upper harmonics sounded sharp and penetrating.

In these experiements, only relative amplitudes could be introduced. Von Helmholtz indicated the amplitude of the various forks by assigning them musical loudness values, such as mf, f, p, etc. This is acceptable in view of the limited instrumentation available at the time. His important conclusion was that the quality of a sound depended upon its power spectrum.

By varying the aperture of the resonance chamber, Von Helmholtz could investigate the effect of phase on timbre perception. He also added lumps of wax to the tongs of the tuning forks so as to put the forks "as much as we please out of tune." He found that the results were the same as when the components of the tone were in phase. He states that "at least there is no alteration so marked as to be recognizable after the expiration of a few seconds.. . . and no such change of quality as would change one vowel to another" (p. 125). Further, he did not consider it impossible that, since harmonics beyond the 6th to 8th gave rise to dissonance and roughness, a phase effect did not exist for these higher harmonics.

Timbre of Vowel Sounds

Hermann (1890) found that pronounced resonance peaks in the partial spectrum contributed to the recognition of tone color; he termed these peaks "formants." In vowel sounds, he noticed that two formants could characterize most sounds. A chart was produced relating various vowel sounds to formant frequency. For example, the vowel /o/ was characterized by resonance in the frequency area of 400 and 600 Hz; the vowel /a/ had resonance peaks in the area of 800 and 1,200 Hz. Certain dark vowel sounds such as /u/ were characterized by only one formant; the second formants were of very slight intensity.

Although formants are characteristic of vocal sounds, such resonance peaks are not characteristic of all musical sounds. The formants of the vocal tone are due to the two lower resonance frequencies of the coupled cavities of the vocal tract. Double reed instruments have an ill-defined spectral enhancement around 450 and 1,100 Hz (Roederer, 1975). Most brass instruments do not have frequency-independent formants (Backus, 1969). In all cases, formants are due to the resonance properties of the sound producing mechanism.

Peterson and Barney (quoted in Winckel, 1967) plotted the frequency of formant 1 against the frequency of formant 2 for vowel sounds of a man, woman, and child. The connection of the points created agreed well with the traditional vowel triangle of Hellwag (1781). This two-dimensional configuration hints at timbre's multidimensional nature. The distances in formant frequencies between the man, woman, and child were about 10 percent.

Potter and Petterson (quoted in Winckel, 1967) charted the time sequence of diphthongs in which every 20 msec a point was entered representing the formant value at that moment. The axes were the two formant frequencies characteristic of the vocal sound. Winckel states that

> Every dot on the chart is characterized by a specific tone color. For use in practical music the grid would have to be logarithmic, analogous to a musical scale. Thus we obtain a <u>meter of tone color</u> which, through division of the chart into diamonds, registers the size of the still barely perceivable step-changes in tone color. (p. 17)

Thus is found the first claim for a threshold for timbre. This threshold is related to the position of the formant frequencies, and since many sounds cannot be related to characteristic formants, it is a relatively useless measure of timbre discrimination. The threshold for timbre perceptibility for vocal tone color was found to be around a semitone (Winckel, 1967). The perceived color of a given vowel shifted when the frequency of a given formant was changed around a semitone. Winckel (1967) claims that

> This was established statistically by asking a number of people when they recognized the point at which one tone color faded into a neighboring tone color. This gives us a metrical field with a certain number of sensitive locations as functions of complex acoustical excitations. The valences - which one can also apply in sound psychology, according to C. Stumpf (1893) - do not lie close together; tone colors do not form a continuum in our perception, as one would tend to assume. (pp. 17-18)

The exact conditions under which the study was taken are not clear. The citing of a source from the late 19th century, and the unclear meaning behind the words "tone-color," "valences," and "barely perceivable" leave doubts as to the validity of the results. The study does suggest a more carefully controlled investigation of the difference

threshold for timbre. The difference threshold related to formant frequency could be important information in the construcion of those instruments which resonate at frequencies irrespective of the fundamental, however it seems more important to investigate the amplitude changes in a formant rather than a change of formant frequency.

The finding that differences between vowels are related to specific peaks in the amplitude pattern raised the interesting question of whether, for the same vowel, the frequenices of these peaks do or do not shift in accordance with the frequency of the fundamental. Slawson (1968) attacked this problem. Vowel-like digitally synthesized sounds were created. The stimuli were presented monaurally by headphone. The second stimuli of each presented pair had the fundamental frequency raised an octave above the fundamental frequency of the first sound. The pitch of the 700 msec stimuli was lowered gradually during the last 450 msec to simulate speech. It was found that the difference in vowel quality estimated by a group of observers could be reduced to a minimum if the lower two formants of the sound with the higher fundamental were raised by about 10 percent (confirming Winckel, 1967).

Repetition of the experiment with similar stimuli but of constant pitch, in which the subjects were instructed to use musical timbre instead of vowel quality as a criterion, did not give a pronounced minimum, but a very flat curve over the range -5% to + 10% shift in formants. From these observations it may be concluded that, for the stimuli employed, vowel quality and timbre are correlated notions, and that timbre depends mainly upon the absolute position of the amplitude pattern along the frequency scale rather than on the relative position with reference to the fundamental frequency.

Since the pioneering studies in vowel timbre, much work has been done with formants. G. Fant (1958) produced a speech synthesizer. The synthesizer consisted of a potentiometer controlling a filter adjustment of the formant frequencies of a complex sound generated by a sawtooth generator. Experience indicated that the variation between two formant frequencies in a simplified model was sufficient to describe the timbre if a third fixed frequency formant was added.

The idea that more than two dimensions are needed to describe the timbre of vocal sounds was further researched by Pols (1970). The stimuli constructed digitally from information obtained during analog to digital analysis of vowel sounds. The signals differed only in their frequency spectra; loudness, onset, duration, and pitch were equalized. The fundamental frequency was 123.5 Hz for all signals. By starting the signals always on or near the zero line, the onset transients were minimized. Eleven vowel sounds were used. Spectral analysis of the sounds was made with reference to the critical bandwidth (Plomp and Mimpen, 1968) by using 1/3 octave bandfilters. The sound pressure levels in 18 bands with the 11 sounds created an 11 x 18 data matrix. The task was to see the minimal number of dimensions required to represent the sounds.

Perceptual analysis was done by means of triadic comparisons. A similarity matrix was produced by having the subject listen to all possible combinations of three-signal groups. He could listen as often as he wanted and in any order to the three signals. Duration of each stimulus was about 405 msec, but the subject could switch to another within that time. He then selected the two stimuli which were the most

similar. Results were analyzed using Kruskal's (1964) multidimensional techniques.

Analysis of the similarity matrix indicated that a three dimensional physical and three dimensional perceptual space had a correlation coefficient (per orthogonal axis) of .992, .972, and .742. These values were improved by matching with a six-dimensional physical space. At least three dimensions were therefore required to specify the timbre of vowel sounds. The excellent correspondence between physical and perceptual space suggests that the subjects used for their judgments information comparable with that present in the multidimensional physical representation of the signals.

In summary: 1. Sounds produced by devices with resonating cavities are often characterized by pronounced peaks in the power spectrum known as formants. 2. Vocal sounds are characterized by two major formants, with a third formant area important to vowel discrimination. 3. Thus, the timbre of vocal sounds, and all sounds with formants, probably can be described in as many dimensions as there are formants. 4. Physical and perceptual spaces in timbre perception with vowels show excellent correlation, supporting Helmholtz's (1877) view that timbre is primarily a function of power spectrum.

In addition to these conclusions, the concept of a difference threshold for timbre was forwarded which points the way for the present study. Unfortunately, the methodology and specific parameters of such a study are not clear. In order to investigate the difference threshold for timbre, the aspects of a complex tone affecting timbre must be understood so that they can be controlled.

Phase Effects on Timbre

Since the mathematical defintion of a complex tone includes the aspect of phase (see Equation 1), the question of how much effect this aspect has on the perception of timbre has been of interest to many psychoacousticians. Zwicker (1952) essentially determined a threshold for phase using amplitude and frequency modulated tones. He presented subjects with amplitude and frequency modulated tones alternated with unmodulated tones. The modulation measure was varied between values for which the unmodulated tones and the modulated tones were clearly dissimilar and values for which the subjects could not hear any difference. Four subjects participated in the experiment. Various loudness levels were used with the carrier frequency at 1,000 Hz.

Modulation threshold was plotted as a function of modulation frequency and modulation depth for both amplitude and frequency modulation. For a modulation frequency greater than 80 Hz the two curves converged. Above about 100 Hz for amplitude modulation and 125 Hz for frequency modulation, the components became separately audible. The superimposed curves above modulation frequencies of 80 Hz indicated that the modulation threshold is determined only by the relative levels of the compenents, independent of loudness level. Critical modulation frequency was about 10 percent that of the carrier frequency above 500 Hz. The data points agreed with about half of the critical bandwidth estimate.

These results indicate that for the conditions involved the auditory system is sensitive to phase if the three components of the stimulus fall within a single critical band. Unfortunately, since the critical bandwidth estimates are based on direct-masking, without the

effect of lateral suppression, the results may be purely coincidental (Plomp, 1976, p. 66).

Goldstein (1967) experimented similarly with four subjects. Three-tone amplitude and frequency modulated stimuli with the same power spectrum were presented to subjects at 60 db SL. Amplitude and frequency modulated tones sounded different for relative modulation frequencies up to about 8 percent. There were no differences above 30 percent. In approaching the latter limit, the stimuli sounded more and more like a complex of three separate tones. Similar correspondence was found to auditory critical bandwidths. Strong phase effects seemed discernable within the band. Experiments by Buunen (1974) indicate that combination tones of the $2f_1 - f_2$ class may be responsible in part for phase effect as well as the overlap in critical bandwidth.

De Boer (1961) noticed that neither periodic variations in amplitude nor a change in timbre were introduced if a group of adjacent harmonics was slightly shifted in frequency. He used a three-tone stimulus with a carrier of $nf + f$ and amplitude modulated by f, creating a group of three harmonics of which the equal phase angles increase. He stated a phase rule: The ear is apparently not sensitive to adding equal phase angles to all harmonics, and is not sensitive to periodic variations in the fine structure.

Licklider (1957), however, used a generator which had control over both the amplitudes and phase angles of the first 16 harmonics. He noticed that a change in phase pattern resulted in almost every case in a detectable difference in timbre. In general, changing the phase of a high-frequency component produced more effect than changing the

phase of a lower-frequency component; the effect was larger for low fundamental frequencies than for high ones. His statement in the appendix of the Journal of the Acoustical Soeiety of America contains no clues as to experimental methodology.

Schroeder (1959) found that, in a 31 component tone, timbre strongly depends upon whether the waveform contains large peaks. Weaker, but still audible, effects occur for phase changes that lead to identical peak factors. Small or no subjective changes were produced by variations of the phase spectrum which left the envelope of the stimulus invariant.

Mathes and Miller (1947) amplitude modulated the middle component of a three-tone stimulus. They found that amplitude modulation gives a roughness sensation which disappears by shifting the phase of the middle component over 90°. At a sound level of 60 db the phase effect appeared to be present up to mudulation frequencies of about 40 percent of the carrier frequency.

Craig and Jeffress (1962) investigated the effect of phase on a two-component tone. A series of tones were presented to subjects in random order. The second tone complex was similar to the first but was 180° out of phase. The components of the tone were at frequencies of 250 Hz and 500 Hz. The amplitude of the first component was a constant 60 db SPL, while the second component was varied in intensity between 3 and 73 db SPL in 10 db steps. At the 1 percent level of significance, subjects could identify differences from 13 db SPL to 73 db SPL between the tones. Differences in pitch and timbre at low and moderate intensities of the second component were noticed, while loudness differences appeared at high intensities.

Bilsen (1968) observed that up to a repeition frequency of 700 Hz, periodic noise can be discriminated from narrow periodic pulses. These stimuli have similar amplitude spectra but very different phase spectra. The frequency limit of the repetition frequency appeared to decrease if a low-pass filter was introduced, thereby confirming the view that phase effects are caused by the higher harmonics.

Raiford and Schubert (1971) presented standard and comparison octave complexes in which the comparison stimuli's phase angle was one of 18 randomonly chosen values from 0° to 345°. The subjects were required to decide whether the standard and comparison stimulus were the same or different. The first component of the stimulus had a fundamental frequency of 250 Hz at 60 db SPL; the second component was at 500 Hz and 50 db SPL.

The results for six subjects showed that the stimuli could be progressively better discriminated as the phase angle deviated more from 0° up to a broad maximum near 180°. Discrimination improved from 50 percent correct at 0° to 95.5 percent correct at 180°.

Plomp and Steeneken (1969) applied the method of triadic comparisons to the problem of phase effects on timbre. In each of two experiments, sets of stimuli with equal loudness and pitch were compared in triads by a group of eight subjects. They could listen repeatedly to the stimuli within each triad in any order, and had to judge which pair was the most similar and which pair was the most dissimilar successively. The stimuli were presented binaurally by means of headphones at 50 db SPL. The first experiment had eight stimuli consisting of all sine terms, all cosine terms, and alterations of sine and cosine

terms, with amplitudes in reciprocal to the harmonic number. The fundamental frequency was 292.4 Hz.

The dissimilarity matrix constructed from the triadic comparisons showed that the greatest dissimilarity was between tones with all sine or cosine terms and tones with alternating sine and cosine terms. In the second experiment, complex stimuli of all sine terms and alternations of sine and consine terms were compared by triadic comparisons to stimuli with random phase spectra. Kruskal's (1964) program MDSCAL provided a three-dimensional best-fitting configuration of points.

It appeared that the interpoint distance between the two non-random stimuli was not exceeded by the interpoint distances to any two other random phase spectra. On the basis of these results the authors concluded that the dissimilarity between a complex tone containing either sine or cosine terms and a tone containing alternate sine and cosine terms represented the maximal effect of phase on timbre.

In an additional experiment, stimuli of all sine terms and alternating sine and cosine terms were given four different spectral envelopes of -4.5, -5.5., -6.5, and -7.5 db SPL per octave. The resulting eight tones were sorted by the method of triadic comparisons. The dissimilarity matrix was analyzed using INDSCAL (Carroll and Chang, 1970) to account for subject's individual differences in weighing several dimensions of a common perceptual space.

Results indicated that the effect on timbre of varying the phase spectrum of a complex tone was small compared with the effect of varying the amplitude spectrum. The effect of phase decreased with increasing fundamental frequency.

These experiments regarding phase effect on timbre demonstrate that: 1. Phase effects are easiest to discern as the phase angle diverges from 0°. 2. Phase effects appear to be greater in the higher harmonics (n greater than 8). 3. Phase effects are discerned more readily if the components are within a critical band. 4. Phase effects are minimal if the same phase angle is added to all components. 5. The effect of phase on timbre is smaller than the effect of varying the amplitude spectrum.

From these conclusions, it is apparent that phase angle cannot be disregarded in generating complex tones for discrimination studies. Stimuli of all sine or cosine terms would appear to be best suited for timbre discrimination studies, and the angles of the components should be inflexible for all stimuli used. Using complexes of small numbers of partials would be advantageous since any imperfections in phase angle would be less likely to distort the results than in complexes with large numbers of partials.

Effect of Amplitude Spectrum on Timbre

At least two investigations previously have researched the effect of varying the amplitude of a single component in a complex sound. Unfortunately, both are of German origin and are unavailable. Lob (quoted in Plomp, 1970, p. 402) investigated timbre changes in complex tones consisting of eight harmonics when the sound pressure level of a single component was altered. No other information could be found concerning this study in the research of the writer. Stumpf (quoted in Winckel, 1967, p. 113) used interference to alter the amplitude of single partials of a complex sound. He found that amplitudes and frequencies of single partials of a sound spectrum could be changed greatly before a

distortion of the tone color was noticed. In fact, the erasing of an entire partial, so long as it was not a formant, would not be noticed by the untrained ear. Increasing the amplitude of an overtone until it was heard by itself also caused a change of tone color. Significant in these statements is the reference to "untrained ears;" the meaning of this, without further explanation, is nebulous. However, the present study should provide evidence from which to judge the results from an experiment conducted by Stumpf over fifty years ago.

Plomp (1970) conducted a definitive investigation into the physical-perceptual components of timbre. The timbre difference between complex tones of equal loudness and pitch were compared in subsets of three stimuli by the triadic comparison technique. A digital computer was used to synthesize the sounds. Each time the subject pressed a button on a triangular box, he heard one of the stimuli of a particular triad. He could listen as often as he wanted and in any order he wanted to the three signals. The duration of each stimulus was maximally 405 msec, but the subject could switch to another signal within that time. When he had selected the two stimuli which, in his opinion, were the most dissimilar, he pressed the response button; he did the same for the most similar pair. Both responses were punched and typed by computer, after which the next group of three stimuli were presented.

The stimuli were derived from musical tones. In one experiment a single period of the same note played by nine musical instruments was used ($\underline{f}_o$ = 319 Hz). In a second experiment, these single periods were

adopted from ten stops of a pipe organ (f_o = 263 Hz). The periods were stored in digital form in the memory of the computer, and could be generated as a continuous signal.

A matrix was constructed in such a way that every time a pair was judged most similar the appropriate cell in the matrix was updated with +2 points; the most dissimilar pair got 0 points, and the cell concerned with the remaining pair got one more point. Ten subjects created the dissimilarity matrix. Correlation coefficients had to be created to compare the dissimilarity indices with the power spectra of the sounds. Kruskal's (1964) multidimensional scaling method was used to accomplish this. To quantify the amplitude pattern, Plomp used 1/3 octave bands in complex sound analysis, using the SPL's in db as a measure of the contributions of the different frequency bands. He states that "it is reasonable to assume that, with the sum of these specific loudness contributions determining the overall loudness of a sound, the shape of their profile determines the timbre" (Plomp, 1976, p. 94).

The sound spectra of the nine stimuli were analyzed with a set of 15 1/3 octave bandfilters with center frequencies of 315 - 8,000 Hz. The output SPL's were considered as coordinates of a point in a multidimensional space with orthogonal axes. Thus there was a 15-dimensional representation of the sound spectra of the stimuli. In order to obtain a single index for each spectrum, the following formula was applied:

$$D_{i,j} = p\sqrt{\sum_{k=1}^{m} (L_{ik} - L_{jk})^2}\ , \qquad (8)$$

with $D_{i,j}$ = difference in frequency spectrum between the tones i and j; $L_{i,k}$ = SPL of tone i in band k; m = total number of frequency bands. Power p = 2 gives the Euclidean solution, implying that timbre dissimilarity is mainly determined by the largest difference in sound pressure level present in any frequency band; power p = 1 gives the area solution, implying that the dissimilarity is mainly determined by the number of frequency bands for which the sound pressure levels of the tones differ significantly, irrespective of how large these differences might be. The power variable p is not known *a priori*, and was treated as an unknown value for which the optimum value that gives the best correlation with the dissimilarity indices could be determined.

Correlation coefficients between .81 and .86 were obtained under the most favorable p values. Correlation between dissimilarity indices and $D_{i,j}$ as a function of p yielded a rather flat curve, suggesting that that the stimuli used were not very appropriate to decide which p-value was actually involved in timbre discrimination. Plomp (1970) concluded that additional investigations would be required using bandfilters with bandwidths more accurately set to the critical bandwidth. He also determined from these high correlation coefficients that the timbre of a given steady state complex and its sound spectrum are clearly related, no matter what p value is chosen. A given sound could be represented by as many as 15 dimensions, but Plomp (1970) found that only 3 dimensions were sufficient to account for the perceptual differences of the stimuli used in his experiment.

Bismarck (1974) investigated the verbal attributes of steady-state complex sounds. Thirty-five tones were used in the investigation,

all but five with a f_o of 200 Hz. Bands of noise and steady-state parts of vowels were used. All stimuli were matched to stimulus 8 which had a SPL of 60 db. The sounds were presented binaurally thru headphones to eight musically trained and eight unmusically trained subjects. The subjects were requested to rate the stimuli individually along 30 carefully selected 7-point semantic scales. From the average ratings of each group of subjects, a correlation matrix was calculated.

Four main factors were statistically determined: hard-soft, compact-scattered, empty-full, colorless-colorful. These accounted for 9 percent of the variance. The sharpness factor increased when either the upper limiting frequency was increased or the slope of the spectral envelope was decreased. The attribute of compactness was used by subjects to differentiate between complex tones and noise. Von Bismarck concluded that sharpness is primarily related to the center of gravity of loudness on a frequency scale in which critical bandwidths have equal lengths.

Grey (1975) also investigated timbre using multidimensional scaling techniques. Stimuli were derived from 16 instrumental tones played near Eb (311 Hz) with durations between 280 and 400 msec. The recorded sounds were digitally analyzed and simplified from 500 to an 8 line function, from which new signals were digitally synthesized. Sampled at 25,000 samples/sec, these were equalized for loudness, pitch, and duration, and were recorded on a Sony tape recorder on Maxell tape at 7.5 ips.

The sound stimuli were played in a room through loudspeakers. Twenty subjects participated in the investigation, with 270 trials to each

subject, 30 of which were practice trials. There were 16 x (16 - 1) = 240 possible pairs of tones given both directions in random order. The subject was told to relate the similarity of the two tones presented relative to all other tones heard. Rating was on a scale of 1 to 30: 1 - 10 was very dissimilar, 11 - 20 was average, and 21 - 30 was very similar. A 16 x 16 matrix of data was analyzed by INDSCAL, a multidimensional data analysis computer program.

Results indicated that three dimensions were most satisfactory to describe the complex tones: 1. Spectral energy distribution (power spectrum); 2. Low amplitude, high-frequency energy in the attack; 3. Synchronicity in the higher harmonics along with the closely related level of spectral fluctuation in the tone through time (fine structure fluctuations).

Several timbre studies have been conducted to investigate the importance of onset characteristics in timbre perception. Both Stumpf (1883) and Elliott (1975) severed the onsets from sounds recorded by various musical instruments. Findings showed that instrumental sound of constant pitch and intensity loses its character if one deletes the attack. Subjects confused a tuning fork with the flute, oboe for clarinet, cello for bassoon, cornet for violin, and French horn for flute in these studies. One must remember that this is a verbal response type study. The subject must respond in terms of a learned verbal text; more information is required to do this, it would seem, than to merely recognize differences in timbre. This conclusion would seem justified by the information presented in this last section about the overwhelming importance of power spectrum in timbre perception. Semantic scales, such as that used by Grey (1976), can yield biases if such scales are not carefully

constructed. Grey's scale of 30 points seems to this writer extremely wide and unwieldly for use over such a large number of trials. More work needs to be done on perceptual-verbal aspects of timbre.

Experiments on the effect of amplitude spectrum on timbre indicate that: 1. Amplitudes and frequencies of single partials of a sound spectrum can be changed greatly before a distortion of the tone color is noticed. Erasing an entire partial, so long as it is not a formant, is not noticed by the "untrained ear". 2. There is a high correlation between sound spectrum (physical space) and timbre (perceptual space). 3. Four main semantic scales were found amoung thirty carefully selected scales to describe different timbres. The attribute of sharpness could be halved and doubled; it was concluded that sharpness is related to the center of gravity of loudness on a frequency scale in which critical bandwidths have equal lengths. 4. Three dimensions were found to be the most satisfactory to describe the timbre of complex sounds: power spectrum, onset, and fine structure fluctuations.

CHAPTER III

METHODS AND MATERIALS

Introduction

The purpose of this investigation is to determine the difference threshold for timbre related to the power spectrum of a complex sound. Prerequisite to determining an experimental methodology for obtaining the difference threshold is a means for controlling the power spectrum. In this investigation, the intensities of each partial of a sound complex must be controlled independently and with great accuracy. Experiments cited in Chapter II used essentially three procedures for obtaining stimuli for use in timbre discrimination: varying the intensity of a given partial by masking it with a probe tone, using phase-locked oscillators, and digitally synthesizing complex waveforms.

The use of masking procedures allows one to eliminate a partial, however one cannot increase the intensity of this partial relative to the rest of the sound complex unless the other partials are partially masked. This procedure is cumbersome at best; any procedure such as this probably would suffer from phase effects. Further, it would be very difficult to attempt to control for loudness variations using masking procedures, because one cannot control the exact amplitude of each partial with sufficient accuracy.

Phase-locked oscillators would control for phase effects. However, if a large number of stimuli are used, this is impractical because each stimulus would require careful adjustment using separate oscillators for each partial. The waveform then would require analysis using frequency counters and sound and vibration analyzers. Further, phase-locked oscillators were not available to the writer. If as many as 350 different stimuli must be generated, as has been projected for the present study, this became impractical with respect to time.

Computer synthesized sounds, therefore, appear to be the best solution. Each parameter of each partial could be carefully controlled through appropriate programming. Separate programs could be created to handle repetitive parameters between stimuli. Once implemented, the entire set of stimuli could be generated in a few days using this program.

Background in Digital Sound Synthesis

Sound is the continuous displacement of particles in an elastic medium, or a continuous fluctuation in pressure in the air. The sound's characteristics are determined by the manner in which the air pressure changes over time. Sounds therefore can be represented by two dimensions: amplitude and time. A graph of these dimensions is called a sound-pressure waveform (Howe, 1975, p. 160).

These sounds are continuous in nature, and are therefore analog signals. Almost all computers are capable of handling only finite numbers, or digital quantities. In order for a computer to produce sound, an analog signal must be represented in digital form by the method of sampling.

Sampling a sound is accomplished by taking the instantaneous amplitude of the sound pressure waveform at successive and equally spaced time intervals. The amplitude factors at each time interval are known as samples, and the number of samples per unit time is known as the sampling rate. If a pure tone of 500 Hz is sampled at a rate of 17,500 samples/sec, each cycle (1/500 sec) of the sine is represented by 35 samples; if the sampling rate is 30,000 samples/sec, each cycle is represented by 60 samples. Thus it is clear that if the sampling time is small, the samples will be a good approximation of the continuous function; if it is large, the approximation will suffer. Of course, the approximation also depends on the type of waveform. More samples are required to approximate a rapidly changing function than a slowly changing one. Mathematically it has been shown that R samples per second are needed to approximate perfectly a function with a bandwidth of R/2 Hz (Mathews, 1969, pp. 12-18). Thus to approximate a sound with a bandwidth of 15,000 Hz, a minimum of 30,000 samples per second are required.

Amplitude of the analog signal is represented by the magnitude of the numbers derived from sampling. This process is known as quantization of the samples. The numbers used for quantization can contain only a limited number of digits, limited by the finite capacity of the computer central processing unit to handle binary numbers. If the numbers in a computer can have only two decimal digits, for example, all the sample amplitudes between 12.5 and 13.5 must be represented by the two-digit number 13. Errors in quantization lead to

amplitude distortion. The magnitude of the distortion is estimated in terms of signal-to-noise ratios. The approximate signal-to-noise ratio in a given digital-processing situation is the maximum number expressible by digits divided by the maximum error in representing any number. If there is a two digit maximum, the maximum number is 99 and the maximum error is .5. The signal-to-noise ratio = 99/.5 = = 200. Since db = 20log amplitude (Howe, 1975), 200 amplitude units equal approximately 46 db. Three decimal digits (ten binary digits) yield a ratio of 66 db, and four provide 78 db. Since most analog tape recorders are not capable of recording sound with a signal-to-noise ratio greater than about 60 db, a computer capable of dealing with 10 binary digits should suffice for generating sound stimuli. Unfortunately, this is not quite correct, since the above signal-to-noise ratios are for pure, notcomplex, sounds. However, since much digital equipment is capable of handling at least 16 bits (yielding a signal-to-noise ratio of greater than 93 db), quantization errors are usually negligible (Mathews, 1969, p. 7).

Each of the dimensions in the original sound pressure waveform is contained in the sampling process: time is represented by the sampling rate, and amplitude is represented by the magnitude of the numbers. If the representation in these two dimensions is sufficiently accurate, a sound can be generated without perceptible distortion. However, there always will be a certain amount of distortion in a digital representation of an analog signal; the limitations of sampling and quantization determine the magnitude of the distortion.

Once the computer has calculated a set of digits representative of a given sound, the digits must be converted to analog form. This is done by a piece of hardware called the digital-to-analog converter (DAC). At the input to the converter, decimal numbers are converted to binary equivalents. For example, the binary number 11011, representing the decimal value 27, is input to the converter. The five binary digits are represented at the converter by five lines going to switch controls. A "1" is represented by a positive voltage and a "0" by a negative voltage. The switch controls close the associated circuits if they have a positive input and open them if they receive a negative input. A resistor network supplied with a reference voltage is attached in matrix to the switching circuits. The voltages sum according to the number of open switches. Thus an analog signal is created representative of the digital input. In an actual converter, the switch controls are flip-flop registers and the switches are transistors. Higher accuracy can be obtained by increasing the number of switching circuits and thus the number of binary digits the DAC can process.

One problem of DACs is inherent in the switching mechanism: If all the switches do not operate at exactly the same speed, large errors will occur briefly during the change from certain digits to adjacent digits. For example, in going from 0111 to 1000 the analog output should change by only one unit. However, all the digits change state. If the most significant digit is slightly faster than the other digits, the actual sequence will become 0111 1111 1000. The analog output would be catastrophic. The regulation of switching speed, timed by a

quartz crystal-referenced electronic clock, is thus of paramount importance. The speed at which the switches operate, approaching 20 to 30k operations/sec, can create electronic impulses near these frequencies in the output signal line from the converter. It is therefore imperative that a band reject filter with a cutoff frequency below the sampling rate, or a bandpass filter with a similar upper-frequency cutoff, be used to eliminate switching transients from the output.

Music V

The digits required for conversion to an analog signal can be provided by two means. The digits can be obtained from an analog signal processed through an analog-to-digital converter, which is essentially the reverse process of digital-to-analog conversion. The second procedure involves having the digital computer provide the required sequence of digits according to a software program. Music V is such a piece of software.

Music V was written by Dr. Max Mathews of Bell Telephone Laboratories, and is described in detail in his book, The Technology of Computer Music (Mathews, 1969). The version used in this investigation was implemented by Roy Campbell (1975) on the Honeywell H635 computer system at The University of Kansas. Further modifications, correcting errors in the original coding were made by the writer in 1976.

Music V was written entirely in FORTRAN, except for one important machine-language subroutine FROUT which outputs samples to magnetic tape before digital-to-analog conversion. Before describing

the use of the program in recipe fashion, it is important to examine the concepts behind the program which allow it to be used to generate sound stimuli with specified parameters.

Music V Organization. Music V consists of three program sections called passes. These sections perform different functions related to reading data and processing information. In general, Pass I is an input routine, Pass II a data organization routine, and Pass III a calculcation and output routine.

The purpose of Pass I is to read the input data and translate it into a form acceptable to the subsequent passes (see Appendix A for a listing of the program with headings and comments). The interpretive input routine READ1 (and READO which reads the first data card and initializes READ1) is written in FORTRAN IV, but requires two machine-language subroutines MOVL and MOVR (not included in the appendix because of their machine-specific nature) for the purposes of character shifting. The input data are provided via IBM cards. Data can be punched in free format without regard to the number of blanks between statments. A data statement always begins with an operation code. Other fields of information in the statement are separated by blanks or commas. Fields separated by successive commas are assumed to have the value 0. A data statement always is terminated with a semicolon. With the exception of certain statements used to define instruments, the fields of data statement are referred to as P fields, since this information is stored sequentially in an array labeled P in the program. The operation code is written as a three-letter mnemonic which is stored in the first

P location P(1). The second field gives the action time in seconds from the start of the program, and specifies when an operation is to be executed. Subsequent field designations are dependent upon the characteristics and requirements of a given operation. The input data are terminated with a data card having an operation code of TERM.

Pass I checks the data statements for errors. If there are errors, termination of the program is accomplished by branching to a non-existent subroutine called HARVEY without proceeding to Pass II. As the data cards are read, they are printed, and any error comments are printed after the inaccurate statement.

Pass I contains a data array D(2000), which is used to store values needed in subsequent processing. Location D(4) is reserved for the value of the sampling rate, which is set by the first data card of a given set of instructions. After the number of fields in a data statement has been established and the P array is appropriately filled without error, Pass I calls upon WRITE1 to write the data on tape or disc so as to be available for Pass II.

Pass II reads data into its arrays by calling READ2, reading an all numerical score as interpreted by READ1 in Pass I. Routine SORT is called, and all the data are sorted into forward chronological order according to start time. Datum manipulations are handled by CONVT routines, which convert certain amplitude and frequency parameters used in Pass III. Pass II then calls WRITE2 to output a report of the processing progress to the printer. This report shows the converted mnemonics of the operation codes in numerical form.

Pass III reads in a sequence of data statements ordered according to increasing action times, and then executes the operations specified by these data statements. The principal operations are defining instruments and playing notes. In addition, functions, variables, and numbers are computed and stored in the Pass III memory for subsequent use in playing notes.

The main loop of Pass III reads data from the I array into the P array. As in the previous passes, the P array is used exclusively for reading and processing the data statements. The operation code appears in P(1) and the action time in P(2). Samples of acoustic output are generated until the played time equals the action time. The operation code then is interpreted and executed. The next data statement then is read and processed.

Pass III calls on routine FORSAM to execute the outputing of acoustic samples to magnetic tape. SAMOUT is called to scale the samples ready to be output, and this routine calls FROUT, which outputs the samples to magnetic tape (FROUT is written in assembly language for speed of execution). Pass III also generates the numbers representative of various waveforms. GEN1 is called to generate functions composed of segments of straight lines, GEN2 creates functions composed of sinusoids, and GEN3 generates functions specified by relative amplitudes at equally spaced points along a continuous periodic function.

Programming with Music V. Music V was written to allow composers flexibility in writing music via the computer. In order to facilitate accomodation to the computer, much of the nomenclature reflects this utilitarian orientation. The various sample-producing subroutines are accessed to form an orchestra consisting of instruments. These instruments consist of distinct components called unit generators, which are analagous to the modules of an electronic music synthesizer. The composer-performer writes a score consisting primarily of two kinds of input statements: notes and functions. Each component of the orchestra is defined by programming via IBM cards which are submitted for processing by a card reader.

It is best to think of these subroutines in terms of their synthesizer counterparts when using Music V for generation of psychoacoustical stimuli. In general, an "instrument" is the connection of an oscillator which has been given specific frequency, amplitude, and waveform parameters to an output box. Data cards provide specific data concerning frequency, amplitude, and waveform, as well as specifying when a particular sound event should occur, and when the entire run of stiumli is over. In the discussion that follows, this simple approach to the Music V program will be used; those wanting more specific and expansive information should consult Mathews (1969).

An instrument definition is given by the statement INS N_1 N_2;, where INS is the three-character operation code, N_i is a real number

defining the moment in time the instrument is defined, and N_2 is an integer representing the number of the instrument. All time values are in seconds from the start of the run. Therefore the statement INS 0 1; would instruct Music V that the cards which follow pertain to the first instrument defined at the start of the run. The data card END; terminates the definition of an instrument.

The most important component of an instrument is the unit generator called an "oscillator." The data card statement is OSC I1, I2, 0, F_j, S; where OSC is the operation code, I1 is amplitude, I2 is an increment value which determines frequency, and F_j is a function input that controls the output 0 of this unit generator. S is a summing variable which adds successive values of the frequency parameter I2. The mathematical algorithm for simulating an oscillator is described by the equation

$$\underline{O}_{\underline{i}} = \underline{a}_{\underline{i}} * \underline{F}(\underline{S}_{\underline{i}} \bmod \underline{FL}) \tag{9}$$

and

$$\underline{S}_{\underline{i}+1} = \underline{S}_{\underline{i}} + \underline{I}_{\underline{i}}$$

where $\underline{O}_{\underline{i}}$ is the $\underline{i}$th output sample, $\underline{A}_{\underline{i}}$ is the $\underline{i}$th amplitude input, $\underline{I}_{\underline{i}}$ is the $\underline{i}$th increment input (which controls frequency), $\underline{F}$ is a stored function, $\underline{S}_{\underline{i}}$ is the $\underline{i}$th sum of increments, and $\underline{FL}$ is the length of the stored function in samples.

$\underline{F}$ is represented in the computer as a block of samples representing a function defined by a GEN statement (which is discussed below). This function is represented by 512 samples. Since one of the samples is at the 0 time position, the sample length is determined to

to be N - 1, where N is the number of samples. In Music V this is 512 - 1 = 511 samples.

The frequency input parameter must be calculated in relation to this function length. The oscillator simply repeats the function over and over. The following derivation can be made to provide a means of calculating values for input parameter I2:

Frequency (Hz) = sampling rate / samples per period

and

Samples per period = function length / increment

therefore

Frequency (Hz) = sampling rate * increment / function length

and

$$\text{increment in samples (I2)} = \frac{\text{function length * frequency}}{\text{sampling rate}}. \quad (10)$$

For a 500 Hz signal to be generated at a sampling rate of 17,500 samples/sec,

I2 = 511 * 500/17,500 = 14.6.

The amplitude parameter I1 is amplitude scaled from 0 to 2,000. The relation of this to decibels is given by the equation

$$I1 = 10.0^{(db/20.0)}. \quad (11)$$

The exact nature of this decibel was derived by the writer and will be presented later.

In writing a data card for an oscillator, the parameters are represented by array locations in the main program rather than by numbers. The numbers are provided by other cards in the deck. Values for I1 and I2 are obtained from the P array beyond P(4). Values for F

are obtained from an F array, and the variable S can be any P array location not in use. The output of the oscillator is directed toward the B array, which serves as an input-output buffer. B1 is reserved to hold the final output of an instrument. A typical data card defining an oscillator would be OSC P5, P6, B2, F2, P30; (parentheses are not used for array subscripts).

Another unit generator is the "output unit." The symbolic representation of this is OUT I, O; where OUT is the operation code, I is a set of samples for input and O is an area committed for summated output. The statement adds the specified input I into the output block O thus combining it with any other input going into that output block. Since the array B is an input-output block, a common data card would be OUT B2, B1;, where B1 is the special block used only for output.

A typical set of cards defining an instrument would appear as follows:

```
INS 0, 1;
OSC P5, P6, B2, F2, P30;
OUT B2, B1;
END;
```

In this instrument, which is the first instrument and is defined at the start of the run, there is one oscillator using P5 for amplitude, P6 for frequency, and F1 for function (waveform). P30 is used to store the sum of I2 (P6) values required by the oscillator, and the output of the oscillator goes to block B2. The output generator OUT takes the output of the oscillator in B2 and places it in special output block B1 where, during Pass III, it is written onto magnetic tape. It should

be noted that the last two cards of an instrument definition are the only ones which do not have an action time as the second field.

Besides defining instruments it is also necessary to define the functions used by the oscillators. This card would normally come after an instrument definition END card. The specific parameters of the GEN statement used to create these functions are dependent upon which of the function generators is called. A typical GEN card would read:

GEN 0 2 1 1 1;.

This calls on function generator 2 at the start of execution. GEN2 produces the sum of sinusoids. The third field states that this will be function 1 (F1); the fourth field states that the fundamental is to be computed with an amplitude of 1, and the fifth field states that there will be one harmonic. Thus the above note card creates a function F1 which is a sinusoid.

To have an instrument play, a NOT card is used. The number of parameters depends on the instrument definition. A typical NOT card would read:

NOT 2.0, 1.0, 2.0, 118.85, 14.6;.

In this example, the first field is the operation code, the second field tells when the note should be played (2 seconds from the start of execution), the third field indicates which instrument shall be played, the fourth field indicates the length of the note, the fifth field is the amplitude input parameter for the oscillator, and the sixth field is the frequency input parameter.

The instruments of an orchestra have the unique property in that they can play as many as 30 different notes at the same time. Thus a single instrument can be defined and a number of different components

can be adjusted with one instrument. If the following card order occurs,

```
NOT 2.0, 1,  2, 118.85, 14.6;
NOT 2.0, 1, 2, 118.85, 29.2;,
```

instrument 1 will produce a two second sound consisting of two components, one at 500 Hz and one at 1,000 Hz, at 41.5 db (refer to Equations 10 and 11).

The first card in the data deck is the card SIA 0 4 N;, where N is the sampling rate in samples/sec. Thus SIA 0 4 17500; would set the sampling rate (set in P(4))to 17,500 samples/sec. With this information, it is possible to provide an example of Music V programming:

```
SIA 0 4 17500;
INS 0 1;
OSC P5, P6, B2, F1, P30;
OUT B2, B1;
END;
GEN 0 2 1 1 1;
NOT 1 1 10 118.85 14.6;
```

Upon execution and digital-to-analog conversion, this data deck would produce a sine wave at 500 Hz with a duration of 10 seconds and a relative amplitude of 41.5 db.

Pilot Studies

A review of the literature revealed that a study of the difference threshold for timbre related to power spectrum had not been reported. Hence there is no model upon which to base the current investigation. The specifics of the values to be given to the amplitude spectrum, and the methodology to employ, therefore were determined by a series of

informal pilot studies.

The first study concerned an analysis of the output of Music V. Three decks of cards, containing the Music V program, were submitted:

```
SIA 0 4 30,000;
INS 0 1;
OSC P5, P6 B2 F1 P30;
OUT B2 B1;
END;
GEN 0 2 1 1 1;
NOT 0 1 2 316.228;
TERM;
```

This program would produce a sine wave with a frequency of 500 Hz and a relative amplitude of 50 db. The digital tape was processed through the Special Media Input-Output System (SMIOS) at The University of Kansas by Pete Herrick (M.S., University of Kansas). The SMIOS system is a digital-to-analog converter with peripheral equipment to be used for specialized digital processing. The output from the SMIOS system was applied to the input of a Radio Shack model ST-120 tape deck at the left channel.

Only qualitative analysis was employed. The resultant output did not sound like a sine wave. There were many fluctuations of pitch and timbre. Hardware investigations revealed that the clock timing the digital-to-analog converter would not hold steady at 30,000 Hz. It was decided to run the same deck with the first card altered to SIA 0 4 17500;, and to lower the digital-to-analog converter clock accordingly.

The qualitative perception of the resultant sound indicated a stable pitch and sonance. The signal then was applied directly to a General Radio type 1192/Z frequency counter, which read a stable 499.998 Hz. The signal then was passed by line through a General Radio model 1564 sound and vibration analyzer. Since the background noise of the equipment was greater than 50 db SPL, the signal was monitored at 80 db SPL. The output remained stable at this level. The signal then was passed into a General Radio oscilloscope. The output signal had the form of a sine wave, but there was evidence of distortion in small peaks along the curvature of the function. Applying a lowpass filter with a cut-off frequency of 5,000 Hz and a slope of 12 db/octave resulted in some smoothing. The distortion in the signal was surmised to be due to sampling error.

The program itself was searched for error. It was found that the oscillator subroutine FORSAM contained a truncating function. Thus amplitude magnitudes were being truncated past the decimal point. The writer reprogrammed the routine with an interpolating oscillator algorithm. The card deck then was submitted and translated.

The output signal on the oscilloscope appeared to have fewer peaks, but some were still present. This sampling inaccuracy was accepted as part of the nature of digital sound processing. The discussion concerning sampling error and signal-to-noise ratio suggested that this was to be expected (Mathews, 1969).

An additional card—NOT 0 1 10 118.85 29.2;—was added to create a second component at 1,000 Hz with a relative amplitude of 41.5 db.

After conversion on the SMIOS, the signal was passed through the lowpass filter and into the sound and vibration analyzer. The background noise from the system was attenuated. With a 1/10 octave bandpass, and centered on 500 Hz, the IL reading was 50 db; with the filter centered on 1,000 Hz, the reading was 41.2 db.

The writer therefore had reasonable confidence in the ability of Music V to produce complex sounds with specific amplitudes and frequencies. A procedure from which to establish a difference threshold then was devised. The method of adjustment was rejected because of the need for real-time sound synthesis. The method of constant stimuli was rejected because of the need for stimuli which exist on a single continuum. The method of limits was chosen because it could be adapted for responses from a 2AFC procedure, and because it did not need real-time sound synthesis.

The standard stimulus chosen was defined by the equation:

$$\underline{p}(\underline{t}) = \sum_{n=1}^{m=6} \underline{a}_n \sin(2\Pi\underline{n}*500*\underline{t} + \Phi), \qquad (12)$$

where $\underline{p}$ = pressure, $\underline{a}$ = amplitude, $\underline{t}$ = time, and Φ = phase angle. This complex sound consists of a fundamental frequency of 500 Hz and 6 partials in integral multiple relation to the fundamental. The fundamental frequency was chosen so that each component of the harmonic complex would be within its own critical band. This facilitated loudness calculations and analysis of the contribution of each component to the overall timbre. The complex is also well within the range of hearing. The stimulus bandwidth is within the 8,750 Hz cut-off of computer synthesis based on a 17,500 Hz sampling rate. The relative phase of each component was 0° to control for possible phase effects.

The first study was largely guess work designed to ascertain the basic level of sensitivity of an individual to timbre changes. The method of limits paradigm was used. The standard stimulus consisted of the complex sound defined in Equation 12 with each partial at 41.5 db IL. The overall sound was therefore 50 db IL. The independent variable was the intensity level of the seventh partial. This was increased by five .1 db steps above, and decreased by .1 db steps below the standard stimulus. The energy gained or lost from the seventh partial was redistributed among the other six partials so that the sound would remain at an overall 49.95 db IL. Table 1 presents the values of the standard and comparison stimuli which Music V used for stimulus generation. The comparison stimuli were arranged from the least amount of energy in the seventh partial to the most (from the bottom of Table 1 to the top). The standard stimulus always preceded the comparison. Each stimulus had a duration of 2 sec followed by 2 sec of silence. Three seconds were allowed for decision making between stimulus pairs. The digital tape was converted on the SMIOS system and placed on standard magnetic recording tape for playback at 7-1/2 ips.

The sounds were presented through Koss Pro/4AAA headphones at 49.9 db SPL to a subject seated in the psychoacoustic laboratory at The University of Kansas. Signals from the tape recorder were passed through a MacIntosh C-26 preamplifier with a lowpass filter at a 5,000 Hz upper cut-off frequency to reduce transients from the digital-to-analog converter.

The subject was told that stimuli would be presented in pairs, the comparison stimulus two seconds after the standard. He then would

have three seconds to decide whether the stimuli were the same or different in timbre. The subject was instructed to pay close attention only to the timbre of the tones, and not to any apparent loudness, pitch, or sonance changes. The subject was told that this was not a test of hearing ability or musicality. The concept behind the generation of the sounds and the methodology employed were fully explained. The subject was instructed to write "S" for same and "D" for different for each pair of sounds, and that these were the only two choices available.

The results were disappointing. No changes in timbre were noticed by the subject using the values in Table 1. All decibel values, except that of the standard stimulus, were calculated and are refered to as "relative decibels" in this paper. Fearful that the generated stimuli did not conform to the input parameters, the sounds were analyzed with 1/10 octave band sweeps of the sound and vibration analyzer. Results indicated that the components of the stimuli were within 1 percent of the prescribed values. It was clear that the energy limits specified in this pilot were too small for a difference threshold for timbre to be determined. The data cards were resubmitted with values of .5 db in change for the seventh partial. Results were still the same, no differences between standard and comparison stimuli.

The design of the methodology was changed slightly for the second pilot. The standard stimulus parameters were the same as in pilot 1. This time, nine values above and nine values below the standard were used for the comparison stimuli. The fourth (middle) partial was varied in this and subsequent pilot studies. Instead of expressing the change of the partials in equal db amounts, a standard increment in absolute intensity was used. The second pilot used an increment of

Table 1

Amplitude Values for Pilot Study #1

Partial 7		Other Partials	
Relative db	IL w/m^2	Relative db	IL w/m^2
42.0	$2.5346*10^{-8}$	41.42	$1.3867*10^{-8}$
41.9	$1.5514*10^{-8}$	41.44	$1.3928*10^{-8}$
41.8	$1.516*10^{-8}$	41.46	$1.39*10^{-8}$
41.7	$1.482*10^{-8}$	41.47	$1.4043*10^{-8}$
41.6	$1.448*10^{-8}$	41.49	$1.4099*10^{-8}$
41.5#	$1.412*10^{-8}$	41.5	$1.412*10^{-8}$
41.4	$1.3833*10^{-8}$	41.52	$1.421*10^{-8}$
41.3	$1.3519*10^{-8}$	41.54	$1.4261*10^{-8}$
41.2	$1.3212*10^{-8}$	41.56	$1.4311*10^{-8}$
41.1	$1.2912*10^{-8}$	41.57	$1.4362*10^{-8}$
41.0	$1.2619*10^{-8}$	41.59	$1.4411*10^{-8}$

Standard Stimulus

1.569×10^{-9} w/m^2. Energy added or subtracted from the fourth partial was redistributed among the other six partials as in pilot 1. Table 2 presents the values of the components in the second pilot. Data cards were prepared with the appropriate frequency and amplitude conversions. The amount of time between comparison and standard stimulus was shortened to 1.5 sec, as it was felt that there was too much image fading with the two second gap of the first pilot. The time between stimulus pairs remained three seconds. Again, comparison stimuli were presented from the lowest energy in partial four to the most, bottom to top on Table 2. The subject responded in the same manner as in pilot study 1; equipment used and procedure were exactly the same.

The series of 19 stimuli were presented to the subject three times. The results indicated a "different" response at 34.996 db each time, however the rest of the stimuli elicited a "same" response. The lower limen of the 49.95 db tone was thus approximately 34.996 db.

The third pilot study used a larger (6.334×10^{-9} w/m^2) increment. The procedure, subject, and materials used were otherwise exactly as in pilot study 2. Table 3 is a chart of the values of the standard and comparison stimuli components.

The series of 19 stimuli were presented to the subjects three times. Results were as follows: Upper threshold - 46.763 db, 47.765 db, 46.763 db. Lower threshold - 37.977 db, 36.727 db, 37.997 db. Therefore the mean lower threshold was 37.574 db CSPL (calculated sound pressure level). The mean upper threshold was 47.097 db CSPL. The interval of uncertainty was calculated according to the following relation (Gescheider, 1976):

Table 2

Amplitude Values for Pilot Study #2

Partial Varied (in relation to others)		Other Partials	
Relative db	IL w/m^2	Relative db	IL w/m^2
44.509	$2.824*10^{-8}$	40.706	$1.177*10^{-8}$
44.26	$2.667*10^{-8}$	40.802	$1.203*10^{-8}$
43.997	$2.510*10^{-8}$	40.895	$1.229*10^{-8}$
43.717	$2.353*10^{-8}$	40.987	$1.255*10^{-8}$
43.417	$2.196*10^{-8}$	41.076	$1.281*10^{-8}$
43.1	$2.04*10^{-8}$	41.164	$1.307*10^{-8}$
42.748	$1.883*10^{-8}$	41.25	$1.334*10^{-8}$
42.37	$1.726*10^{-8}$	41.334	$1.36*10^{-8}$
41.956	$1.569*10^{-8}$	41.417	$1.386*10^{-8}$
41.5#	$1.412*10^{-8}$	41.5	$1.412*10^{-8}$
40.987	$1.255*10^{-8}$	41.578	$1.438*10^{-8}$
40.407	$1.098*10^{-8}$	41.656	$1.464*10^{-8}$
39.737	$9.413*10^{-9}$	41.733	$1.490*10^{-8}$
38.946	$7.844*10^{-9}$	41.809	$1.576*10^{-8}$
37.977	$6.276*10^{-9}$	41.883	$1.543*10^{-8}$
36.727	$4.707*10^{-9}$	41.956	$1.569*10^{-8}$
34.966	$3.138*10^{-9}$	42.028	$1.595*10^{-8}$
31.956	$1.569*10^{-9}$	42.098	$1.621*10^{-8}$
0	0	42.168	$1.647*10^{-8}$

Standard Stimulus

Table 3

Amplitude Values for Pilot Study #3

Partial Varied (in relation to others) Relative db	IL w/m^2	Other Partials Relative db	IL w/m^2
49.5	$8.913*10^{-8}$	32.092	$1.619*10^{-9}$
48.751	$7.50*10^{-8}$	37.783	$3.608*10^{-9}$
48.601	$7.246*10^{-8}$	36.432	$4.397*10^{-9}$
48.070	$6.412*10^{-8}$	37.624	$5.786*10^{-9}$
47.465	$5.579*10^{-8}$	38.558	$7.175*10^{-9}$
46.763	$4.746*10^{-8}$	39.327	$8.564*10^{-9}$
45.924	$3.912*10^{-8}$	39.98	$9.953*10^{-9}$
44.884	$3.079*10^{-8}$	40.547	$1.134*10^{-8}$
43.513	$2.245*10^{-8}$	41.049	$1.273*10^{-8}$
41.5#	$1.412*10^{-8}$	41.5	$1.412*10^{-8}$
40.987	$1.569*10^{-8}$	41.578	$1.438*10^{-8}$
40.407	$3.138*10^{-9}$	41.656	$1.464*10^{-8}$
39.737	$4.707*10^{-9}$	41.733	$1.490*10^{-8}$
38.946	$6.276*10^{-9}$	41.809	$1.516*10^{-8}$
37.977	$7.844*10^{-9}$	41.883	$1.543*10^{-8}$
36.727	$9.413*10^{-9}$	41.956	$1.569*10^{-8}$
34.966	$1.098*10^{-10}$	42.028	$1.595*10^{-8}$
31.956	$1.255*10^{-10}$	42.098	$1.621*10^{-8}$
0	0	42.168	$1.647*10^{-8}$

Standard Stimulus

IU = mean upper threshold - mean lower threshold (13)

where IU is the interval of uncertainty. For the above data this value was 47.097 - 37.554 = 9.55 db 6SPL. The difference threshold was calculated using the following relation (Gescheider, 1976):

DT = ½(IU), (14)

where DT is the difference threshold and IU is the interval of uncertainty. For the above data this was ½(9.55) = 4.775 db CSPL. The point of subjective equality was calculated using the relation (Gescheider, 1976):

PSE = ½(mean upper threshold + mean lower threshold) (15)

werhe PSE is the point of subjective equality. For the above data, this value was (47.097 + 37.554) / 2 = 42.32 db CSPL.

The nature of the decibel being used in computer calculations was, to this point, uncertain. The input parameter I1 was calculated using the conversion porvided by Equation 11. However, the actual values for I1 seemed to be linear in nature, scaled from 0 to 2,000. To test this theory, a new set of parameters for sound generation were created. The partial varied was still the fourth, and the procedure was exactly the same as in pilot study 3. This time, however, the increment was expressed in equal values of computer amplitude units (CAUs). Thirty CAU's were used as an increment, with the other CAUs distributed equally among the remaining six partials. Table 4 gives an abbreviated set of parameter values.

Table 4

Amplitude Values for Pilot Study #4

Partial Varied (in relation to others) Computer Amplitude Units	Other Partials Computer Amplitude Units
570	255
540	260
510	270
480	270
450	285
420	280
390	285
360	290
330	295
300#	300
270	305
240	310
210	315
180	320
150	325
120	330
90	335
60	340
30	345

Standard Stimulus

The set of cards was converted by Music V to sample output from SMIOS to audio tape. This was reviewed by the writer at the computer center, and it was immediately clear that this procedure of treating the CAU as a linear quantity would not work. The computer output contained an error message that indicated that values were being generated inside the Music V program that exceeded the available dimensions of the arrays. This would only happen if incorrect parameters were fed into the program.

The decibel used by Music V should be a model of the decibel measured in the "real world." Relative amplitude levels should be indicated by the use of this decibel, so that when a sound with a level of 50 db SPL is specified, one can be obtained by adjustment of the output level from the tape. That this is the case had to be demonstrated by deductive reasoning through mathematical manipulations. This involved relating SPL, IL, and the CAU.

Equation 3 states the relation of power (I) to pressure and the characteristic impedence of the air. Since the threshold of hearing (on the average) = .002 dyne/cm^2 = 10^{-16} w/cm^2,

$$10^{-16} \text{ w/cm}^2 = (.0002 \text{ dynes/cm}^2)^2 \;/\; \underline{p_o c} \text{ and}$$

$$\underline{p_o c} = 4 \times 10^{-8} \text{ dynes/cm}^2 \;/\; 10^{-16} \text{ w/cm}^2, \text{ therefore}$$

$$\underline{p_o c} = 4 \times 10^{8}.$$

This value for the characteristic impedence of the air was used to relate db IL and db SPL, which are equivalent at a specific value, but have different reference bases. To do this, a value of 40 db was plugged into the equations arbitrarily:

$$40 \text{ db IL} = ? \text{ w/cm}^2.$$

From Equation 4,

$$? \text{ w/cm}^2 = 10^{**}(40/10+(\log_{10}10^{-16} \text{ w/cm}^2))$$

$$? \text{ w/cm}^2 = 10^{-12} \text{ w/cm}^2 .$$

From Equation 5,

$$40 \text{ db SPL} = ? \text{ dynes/cm}^2,$$

$$? \text{ dynes/cm}^2 = 10^{**}(40/20 + (\log 2 * 10^{-5} \text{ dynes/cm}^2))$$

$$? \text{ dynes/cm}^2 = 2 \times 10^{-2} \text{ dynes/cm}^2.$$

To check the equivalency of db SPL and db IL:

$$\text{IL w/cm}^2 = (2 \times 10^{-2})^2 \quad / \quad 4 \times 10^8,$$

as established by the above relations. IL therefore equals 1×10^{-12} w/cm^2, showing the equivalency of the above expressions; 40 db SPL = 40 db IL. To relate this to computer amplitude units:

$$\text{CAU} = 10^{**}(\text{db}/20),$$

therefore,

$$\text{db} = 20\log\text{CAU}.$$

So a computer amplitude is a ratio obviously representative of p/p_{ref} in Equation 5. If

$$1\text{CAU} = ?/.0002 \text{ dyne/cm}^2, \text{ then}$$

$$\text{db} = 20\log/?.0002 \text{ dyne/cm}^2.$$

For db SPL = 1, ? = .0002 dyne/cm^2, therefore

$$1 \text{ CAU} = .0002/.0002 \text{ dyne/cm}^2, \text{ and}$$

$$1 \text{ CAU} = 20 \log 1 = 0 \text{ db SPL}.$$

Therefore, a computer amplitude unit is a ratio equivalent in the computer to p/p_{ref} in Equation 5. All linear calculations using intensity in w/cm^2 should be converted to db SPL, and then to CAUs.

The fifth study returned to the use of intensity increments instead of computer amplitude unit increments. In the fifth and subsequent studies, the overall intensity of the comparison and standard stimuli were changed to 50 db SPL to facilitate analysis by the sound and vibration analyzer. The resolution of the experimental paradigm was increased by taking the area of subjective equality found in the third study and omitting part of it. In the fifth study, the intensity increments were $+4 \times 10^{-13}$ w/cm^2 and -1.2×10^{-13} w/cm^2. Three steps were skipped in each direction. Table 5 gives the values calculated to 13 significant digits, and rounded to 10 digits (the large number of digits calculated was required due to the high powers of 10 to which the numbers were taken). Since the input parameter I2 of the Music V program only accepts three decimal places, final rounding to three places was performed on computer amplitude unit calculations. The standard stimulus was the same harmonic stimulus specified in the first pilot study (Equation 12).

The presentation procedure was different for this pilot study. Verbal reports of the subject in pilot study 3 indicated that he was influenced by knowledge of the direction of the stimuli. Further, the subject always knew that the standard stimulus was first. These expectation factors were reported by Gescheider (1976). To compensate for this limitation of the method of limits paradigm, the stimuli were ordered randomly. The comparison stimuli were ordered according to a random list of numbers from one to nineteen generated by a mathematical algorithm on a Texas Instruments model SR-56 calculator. Whether the comparison or standard stimulus was presented first was determined by random generation of numbers between 0 and 1; numbers

less than .5 specified the comparison stimulus was to be presented first and numbers .5 and above specified that standard stimulus was to be presented first. The data deck submitted for Music V processing specified this random ordering, with 1.5 sec between members of a stimulus pair and 3 sec between pairs of stimuli. The response recording and result calculation procedures were the same as for pilot study 3.

Two subjects participated in the experiment. No training was given to either subject, but the concepts behind the study were explained. One subject was a teacher of psychoacoustics at The University of Kansas; the other was a graduate study in music therapy. Each subject was given only one trial. Table 6 gives the responses of both subjects. It was apparent that there was some inconsistency in response. It was hypothesized that elements of the theory of signal detection were being evidenced. On a given trial, there was a certain probability that even with a difference in timbre the subject, due to a number of variables, responded that they were the same, or vice-versa. To compensate for this possibility, a criterion figure was employed in later trials. The difference threshold would be the point that responses changed from one state to another with at least two responses in the new direction. Thus an isolated "erroneous" response in subsequent investigations was ignored.

Using the above criterion, the difference threshold for subject 1 was 3.97db CSPL;for subject 2 the difference threshold was 3.49 db CSPL.

Table 5 ##

Amplitude Values for Pilot Study #5

Partial Varied (in relation to others)			Other Partial		
Relative db	IL w/cm^2	CAU$	Relative db	IL w/cm^2	CAU
47.656	$5.829*10^{-12}$	241.424	38.421	$6.952*10^{-13}$	83.381
47.347	$5.429*10^{-12}$	238.993	38.819	$7.619*10^{-13}$	87.287
47.014	$5.029*10^{-12}$	224.245	39.183	$8.285*10^{-13}$	91.026
46.654	$4.629*10^{-12}$	215.141	39.519	$8.952*10^{-13}$	94.617
46.262	$4.229*10^{-12}$	205.635	39.821	$9.619*10^{-13}$	98.077
45.830	$3.829*10^{-12}$	195.667	40.122	$1.028*10^{-12}$	101.418
45.351	$3.429*10^{-12}$	185.164	40.393	$1.095*10^{-12}$	104.654
44.812	$3.029*10^{-12}$	174.028	40.652	$1.162*10^{-12}$	107.792
44.197	$2.629*10^{-12}$	162.129	40.894	$1.228*10^{-12}$	110.841
41.549&&&	$1.429*10^{-12}$	119.523	41.549	$1.429*10^{-12}$	119.50
40.288	$1.069*10^{-12}$	103.372	41.728	$1.488*10^{-12}$	122.007
39.771	$9.486*10^{-13}$	97.395	41.786	$1.508*10^{-12}$	122.824
39.183	$8.286*10^{-13}$	91.026	41.843	$1.558*10^{-12}$	123.635
38.504	$7.086*10^{-13}$	84.177	41.899	$1.548*10^{-12}$	124.442
37.698	$5.886*10^{-13}$	76.817	41.955	$1.568*10^{-12}$	125.243
36.708	$4.686*10^{-13}$	68.254	42.010	$1.508*10^{-12}$	126.038
35.423	$3.486*10^{-13}$	59.040	42.064	$1.608*10^{-12}$	126.829
33.59	$2.857*10^{-13}$	47.809	42.118	$1.628*10^{-12}$	127.615
30.337	$1.086*10^{-13}$	37.950	42.218	$1.648*10^{-12}$	128.396

Values calculated to 13 places, rounded for this table.

$ Computer Amplitude Units

&&& Standard Stimulus

Table 6

Responses in Pilot Study #5

Pair #	Subject 1	Subject 2
1	D#	D
2	D	D
3	D	D
4	D	D
5	D	S
6	D	S
7	S	S
8	D	S
9	S	S
10$$	S	S
11	D	S
12	S	D
13	S	D
14	D	S
15	S	D
16	S	D
17	S	D
18	S	D
19	S	D

D = different, S = Same

$$ Two standard Stimuli.

Pilot studies 6, 7, and 8 were efforts to increase the resolution of the paradigm where energy is subtracted from the variable partial. Study 6 used increments of 9×10^{-14} w/cm^2. Seven steps were included between the standard and first comparison. Table 7 gives the specifications for the stimuli in this study. Study 7 used increments of 7×10^{-14} w/cm^2, with eleven steps between the standard and first comparison stimulus. Table 8 gives the specifications for study 7. Study 8 used increments of 2×10^{-14} w/cm^2. In these studies, the difference threshold was only determined for the lower values of the fourth partial. Other aspects of the generation, presentation, and procedure were the same as with pilot study 5.

Results from investigation 6 indicated a mean lower threshold of 37.57 db SPL. This threshold was given credence by study 7, but the resolution of study 7 was not any better. Investigation 8 indicated a mean lower threshold of 36.945 db CSPL. From these values it was possible to determine a satisfactory set of parameters for the final investigation.

Final Procedure and Investigation

Using the pilot studies as a guide, an investigation was conducted to determine the difference thresholds related to the alteration of each partial of a seven-component complex sound. The modified method of limits procedure developed in the pilot studies was employed. The specific amplitude parameters of the complex standard and comparison stimuli are presented in Table 9. The standard stimulus was a seven-component harmonic complex with a fundamental frequency of 500 Hz and a relative amplitude of 50 db, as defined by Equation 12. The phase

Table 7##
Amplitude Values for Pilot Study #6

Partial Varied (in relation to others)			Other Partial		
Relative db	IL w/cm^2	CAU$	Relative db	IL w/cm^2	CAU
47.656	$5.829*10^{-12}$	241.424	38.421	$6.952*10^{-13}$	83.381
47.347	$5.429*10^{-12}$	232.993	38.819	$7.619*10^{-13}$	87.287
47.014	$5.029*10^{-12}$	224.245	39.183	$8.283*10^{-13}$	91.026
46.654	$4.629*10^{-12}$	215.141	37.519	$8.952*10^{-13}$	94.617
46.262	$4.229*10^{-12}$	205.635	39.831	$9.619*10^{-13}$	98.077
45.830	$3.829*10^{-12}$	195.667	40.122	$1.028*10^{-12}$	101.418
45.351	$3.429*10^{-12}$	185.164	40.393	$1.095*10^{-12}$	104.654
44.816	$3.029*10^{-12}$	174.028	40.652	$1.162*10^{-12}$	107.792
44.197	$2.629*10^{-12}$	162.129	40.894	$1.228*10^{-12}$	110.841
41.542&&&	$1.429*10^{-12}$	119.523	14.549	$1.429*10^{-12}$	119.523
39.023	$7.986*10^{-13}$	89.363	41.857	$1.534*10^{-12}$	123.837
38.504	$7.086*10^{-13}$	84.177	14.899	$1.548*10^{-12}$	124.442
37.914	$6.186*10^{-13}$	78.649	41.941	$1.564*10^{-12}$	125.043
37.231	$5.286*10^{-13}$	72.703	41.983	$1.578*10^{-12}$	125.641
36.426	$4.386*10^{-13}$	66.225	42.034	$1.594*10^{-12}$	126.232
35.423	$3.486*10^{-13}$	59.04	42.064	$1.608*10^{-12}$	126.829
34.126	$2.586*10^{-13}$	50.85	42.105	$1.624*10^{-12}$	127.419
32.268	$1.686*10^{-13}$	41.057	42.145	$1.638*10^{-12}$	128.007
28.953	$7.857*10^{-14}$	28.031	42.184	$1.654*10^{-12}$	128.591

Values calculated to 13 places, rounded for this table.

$ Computer Amplitude Units

&&& Standard Stimulus

Table 8##

Amplitude Values for Pilot Study #7

Partial Varied (in relation to others)			Other Partial		
Relative db	IL w/cm^2	CAU$	Relative db	IL w/cm^2	CAU
47.656	$5.829*10^{-12}$	241.424	38.421	$6.952*10^{-13}$	83.381
47.347	$5.429*10^{-12}$	232.993	38.819	$7.619*10^{-13}$	87.287
47.014	$5.029*10^{-12}$	224.245	39.183	$8.283*10^{-13}$	91.026
46.654	$4.629*10^{-12}$	215.141	39.519	$8.952*10^{-13}$	94.617
46.262	$4.229*10^{-12}$	205.635	39.831	$9.619*10^{-13}$	98.077
45.830	$3.829*10^{-12}$	195.667	40.122	$1.028*10^{-12}$	101.418
44.351	$3.429*10^{-12}$	185.164	40.393	$1.095*10^{-12}$	104.654
44.816	$3.029*10^{-12}$	174.028	40.652	$1.162*10^{-12}$	107.792
41.542&&&	$1.429*10^{-12}$	119.523	41.549	$1.429*10^{-12}$	119.523
37.698	$5.886*10^{-13}$	76.716	41.955	$1.568*10^{-12}$	125.243
37.148	$5.186*10^{-13}$	72.012	41.987	$1.580*10^{-12}$	125.708
36.518	$4.486*10^{-13}$	66.976	42.019	$1.592*10^{-12}$	126.171
35.784	$3.786*10^{-13}$	61.528	42.050	$1.604*10^{-12}$	126.632
34.894	$3.086*10^{-13}$	55.549	47.082	$1.615*10^{-12}$	127.012
33.776	$2.386*10^{-13}$	48.844	42.114	$1.627*10^{-12}$	127.55
32.268	$1.686*10^{-13}$	41.057	42.145	$1.638*10^{-12}$	128.007
29.938	$9.857*10^{-14}$	31.396	42.175	$1.650*10^{-12}$	128.462
24.559	$2.857*10^{-14}$	16.903	42.206	$1.662*10^{-12}$	128.915

Values calculated to 13 places, rounded for this table.

$ Computer Amplitude Units

&&& Standard Stimulus

Table 9##

Amplitude Values for Final Investigation

Partial Varied (in relation to others)			Other Partial		
Relative db	IL w/cm^2	CAU$	Relative db	IL w/cm^2	CAU
47.014	$5.029*10^{-12}$	224.245	34.183	$8.283*10^{-13}$	91.026
46.654	$4.629*10^{-12}$	215.141	39.519	$8.952*10^{-13}$	94.617
46.262	$4.229*10^{-12}$	205.635	39.831	$9.619*10^{-13}$	98.077
45.830	$3.829*10^{-12}$	195.667	40.122	$1.028*10^{-12}$	101.418
45.351	$3.429*10^{-12}$	185.164	40.393	$1.095*10^{-12}$	104.654
44.816	$3.029*10^{-12}$	174.028	40.652	$1.162*10^{-12}$	107.272
49.197	$2.629*10^{-12}$	162.129	40.894	$1.228*10^{-12}$	110.841
43.480	$2.286*10^{-12}$	149.284	41.235	$1.295*10^{-12}$	113.808
42.621	$1.828*10^{-12}$	135.225	14.341	$1.362*10^{-12}$	116.701
41.549&&&	$1.428*10^{-12}$	119.523	14.549	$1.428*10^{-12}$	119.523
37.231	$5.286*10^{-13}$	72.703	41.983	$1.576*10^{-12}$	125.641
36.661	$4.636*10^{-13}$	68.086	42.012	$1.589*10^{-12}$	126.072
36.005	$3.986*10^{-13}$	63.132	42.042	$1.600*10^{-12}$	126.501
35.232	$3.336*10^{-13}$	57.756	42.071	$1.611*10^{-12}$	126.928
34.291	$2.686*10^{-13}$	51.824	42.100	$1.622*10^{-12}$	127.354
33.087	$2.036*10^{-13}$	45.119	42.129	$1.633*10^{-12}$	127.779
31.417	$1.386*10^{-13}$	37.225	42.158	$1.644*10^{-12}$	128.202
28.667	$7.357*10^{-14}$	27.124	42.186	$1.654*10^{-12}$	128.624
19.331	$8.571*10^{-15}$	9.258	42.215	$1.665*10^{-12}$	129.044

Values calculated to 13 places, rounded for this table.

$ Computer Amplitude Units

&&& Standard Stimulus

of the components was at 0°.

The data cards submitted to the Music V system were prepared by the program found in Appendix C. This FORTRAN program automatically punched cards with the values indicated in Table 9. The order of the comparison stimuli and the order of presentation were randomly determined as in pilot study 5. There were three separate random orders for each of the seven partials used as independent variables, making 21 sets of data cards, and 21 runs of the Music V program. An example of Music V output report using the valences of the study is found in Appendix B.

Each member of the stimulus pair was presented for two seconds, with one second of silence between members of the pair. Four seconds of silence were allowed between pairs of stimuli for the response of the subject. The report in Appendix B shows the programming for producing these conditions with variance of the fifth partial.

The digital tape produced by the Music V program was converted on the SMIOS system with a 17,500 samples/sec sampling rate. The audio tape was played back through a MacIntosh model C-26 stereo preamplifier into model TDH49-10 Z headphones commonly used for audiometric examinations. The left headphone was used to present the stimuli; the right headphone was a "dummy." The left headphone was acoustically coupled to a General Radio model 1564 sound and vibration analyzer to monitor the standard stimulus. A lowpass filter with a 30 db/octave cut-off at 5,000 Hz was placed between the preamplifier and the headphones to eliminate transients from the digital-to-analog converter. The measurement sessions were conducted in the psychoacoustic laboratory at The University of Kansas.

The original design of the experiment called for having the sound stimuli presented at 50 db SPL. Measurements of the ambient sound level in the psychoacoustics laboratory indicated a noise level of approximately 63 db SPL. Numerous attempts to lessen the ambient noise by shutting off electrical devices and closing hall doors did not lower this reading. Measurements using A, B, and C weighings indicated that much of the ambient sound was below 100 Hz; probably induced by the electrical system of the building. In order for an accurate measurement to be made, the B weighing on the sound and vibration analyzer was employed, and the acoustic output of the headphones was adjusted to 70 db SPL.

Subjects were solicited from music students and music camp participants at The University of Kansas. Four music camp participants, one undergraduate music major, and one graduate music major were employed. Three additional subjects, two graduate music majors and a professor of music education, were employed to listen for loudness thresholds; they did not participate in the timbre investigation. Each of the six subjects used in the timbre investigation participated in a training exercise and seven measurement sessions over a two consecutive day period.

The training session consisted of an explanation of the investigation, a demonstration of the methodology employed, and some practice trials. The explanatory part of the session was standardized by the use of the BASIC computer program found in Appendix D. This program was run on a TRS-80 microcomputer system with graphics display. The program explained that timbre is, provided a graphic demonstration of the concept of complex sound generation, and explained the experimental procedure. The subject proceeded at his or her own rate through this

tutorial. The experimenter then answered any questions before the practice trials began.

The subject was told that pairs of stimuli would be presented. One second would elapse between each member of the pair. Four seconds were allowed between pairs to decide whether the stimili had the same or different timbre. The TRS-80 computer was used to record the responses of the subjects. The subject pressed the "S" button on the left side of the keyboard to record a "same" response. A "D" button was arranged on the right side of the keyboard for recording "different" responses. If the subject decided to change a response within the four-second silence between pairs, he pressed the space bar; this allowed him to change the most recent response. The program kept track of which stimulus pair was being presented, and ordered the responses sequentially out of the random order for facilitation of the calculation of the difference threshold. The experimenter was able to recall the responses in sequential order so that the consistency of the responses could be viewed as a training exercise.

For each of the six measurement sessions the subject responded to seven sets of stimuli, one for each partial varied in the seven-component complex tone. Responses were collected by the computer arrangement described above. Each set of stimuli was a random ordering of the 19 comparison stimuli with random presentation of the standard stimulus in paris. Therefore, each partial varied in relation to the rest of the complex produced six sets of data for each subject. The data were averaged, and the difference thresholds were calculated according to the formulas in pilot study 3.

During the last measurement session, the subject was asked to listen for loudness, not timbre, differences between standard and comparison stimuli. One of the random tapes of stimuli for the third partial was played, and the responses were recorded by the computer arrangement. Three additional subjects participated only in this last measurement session; this was preceeded by a training session similar to the one described above.

The results involved calculating the mean difference threshold for timbre for each of the seven conditions for each subject. The mean difference threshold for each partial for all subjects averaged was then calculated. To test whether the difference thresholds obtained for different partials were statistically different, a repeated measurements Analysis of Variance (ANOVA) was calculated with the partial varied as the treatment condition. The specifics of these calculations, as well as additional data, are described in the next chapter.

CHAPTER IV

RESULTS

Quantitative Data

The raw data from the final investigation consisted of the sequences of "same" and "different" responses given by each subject to each series of 19 stimulus pairs. These randomly ordered responses were sequentially ordered according to the values found in Table 9 by the computer program in Appendix D. The transitional points were the intensity values in db CSPL where the "same" responses changed to "different," with at least two responses in the new direction. The upper and lower thresholds were midpoints between these transitional points and the next stimulus value in each direction. The difference thresholds for each subject under seven different stimulus conditions were calculated using Equations 13 and 14 in Pilot Study 3. These thresholds are summarized in Table 10.

There appears to be a substantial difference between subjects in their sensitivity to timbre as related to the power spectrum of these complex stimuli (Equation 12), however there is no statistical test to substantiate this opinion (Furguson, 1976). To test for differences among the means of the seven partial conditions, a treatment by subject analysis of variance was conducted with the partial varied as the independent variable. The dependent variable consisted of the difference

thresholds in db CSPL. The results, presented in Table 11, yield an F ratio of 2.28, not significant at the .05 level. Therefore, the null hypothesis of no difference among the column means is retained: There was not a significant difference among difference thresholds obtained by varying each of the seven components of a complex sound (when such a complex is defined as in Equation 12). The grand mean difference threshold of 4.28 db CSPL is thus an index of discriminatability for this seven component tone. Adding or subtracting 4.28 db CSPL from a component of a standard stimulus defined by Equation 12 yields, on the average, a noticeable change in timbre 50 percent of the time. These data answer questions 1 and 2 under "Research Questions" in Chapter 1.

Since a computer program easily was constructed to conduct Schéffé comparisons, a set of orthogonal comparisons was tested using the data from Table 10. The F ratios from these comparisons are presented in Table 12. It is curious to note that, although the overall ANOVA was not significant, three of the comparisons yielded significant F ratios. It is clear that $\mu 2$ and $\mu 7$ are not significantly different from one another, but are significantly different from the other five means. $\mu 2$ and $\mu 7$ are the smallest column means, however, and they are the most divergent from the grand mean.

In addition to data concerning the difference threshold for timbre, data also were analyzed concerning the upper and lower thresholds obtained for the six subjects under the seven conditions. These data are summarized in Table 13. The average upper threshold for the seven component tone defined by Equation 12 was 65.1 db CSPL; the lower

Table 10

Difference Thresholds from Final Investigation

Subject	Partial* 1	2	3	4	5	6	7
1	5.03**	3.15	2.90	4.48	4.64	4.55	3.43
2	3.90	3.34	3.61	5.14	5.41	5.89	4.76
3	5.23	4.74	5.32	5.41	4.79	4.96	3.68
4	4.70	4.91	5.19	3.94	3.66	4.80	3.31
5	3.32	3.31	3.04	4.45	3.52	4.09	3.19
6	4.47	3.44	4.86	4.78	3.80	4.52	4.33

Column Mean = 4.44

Grand Mean = 4.28

* Partial varied in relation to others.

** Values are thresholds in db CSPL.

Table 11

Treatment x Subjects ANOVA: Data from Table 10

Source of Variation	Sums of Squares	Degree of Freedom	Mean Square	F Ratio
Subjects	7.24	5	1.45	
Partials	5.72	6	.95	2.28
Error (treatment x subject)	12.58	3Φ	.42	
Total	25.54	35		

threshold was 56.5 db CSPL. These averages are made assuming no significant differences among the columns of upper and lower thresholds in Table 13. Treatment by subject ANOVA was conducted to confirm this. Tables 14 and 15 display the results (using the data of Table 13); no significant differences occurred between treatments for either upper or lower threshold data.

An estimate of the strength of association between independent and dependent variables in the treatment by subject design was conducted with the data of Table 11. This $\hat{\omega}^2$ statistic is defined by the equation

$$\hat{\omega}^2 = \frac{\text{SScol} - (K-1)\ \text{MSres}}{\text{SS total} + \text{MSres}} ,$$

where SScol is the sum of squares for the columns, SStotal is the sum of squares, MSres is the mean square residual, and K is the number of columns (Kirk, 1971). Applied to the data of Table 11 this is

$$\hat{\omega}^2 = \frac{5.72 - (6)\ .42}{2554 + .42} = .12$$

The figure of .12 indicates only about a 12 percent association between partial number and the variance in difference thresholds. This is to be expected since the means of the column effects in the difference threshold ANOVA were not significantly different.

There are no data to report from the loudness discrimination part of the study. Both the subjects who participated in the timbre discrimination investigation and those subjects who only participated in the loudness discrimination investigation, could not respond consistently to loudness changes of sounds with the third partial varied in relation to the other six partials (standard defined by Equation

Table 12

Scheffé Comparisons: Data from Table 11

Hypothesis	$F_{6,\ 30}$ Ratio (Scheffe)
$\mu_1 = \mu_2 + \mu_4 + \mu_5 + \mu_6 + \mu_7$	.37
$\mu_2 = \mu_3 + \mu_4 + \mu_5 + \mu_{61} + \mu_7$	3.39*
$\mu_3 = \mu_4 + \mu_5 + \mu_6 + \mu_7$	.68
$\mu_4 = \mu_5 + \mu_6 + \mu_7$	1.79
$\mu_5 = \mu_6 + \mu_7$	.001
$\mu_6 = \mu_7$	7.42*
$\mu_2 + \mu_7 = \mu_1 + \mu_3 + \mu_4 + \mu_5 + \mu_6$	9.48*
$\mu_2 = \mu_7$	.0071

*Significant at .05 level

Table 13

Mean Upper and Lower Thresholds for Timbre

	Partial+													
	1		2		3		4		5		6		7	
Subject	UT#	LT$	UT	LT	UT	LT	UT	LT	UT	LT	UT	LT	UT	LT
1	65.9***	55.8	64.1	57.8	64.4	58.6	65.2	56.3	65.1	55.8	65.2	56.1	63.8	56.9
2	65.2	57.4	65.1	58.4	65.1	57.8	65.6	55.3	64.9	54.0	66.2	54.4	65.6	56.7
3	64.9	54.4	65.7	56.2	65.4	54.8	65.5	54.4	65.2	55.6	65.2	55.3	65.2	57.8
4	65.4	56.0	65.2	55.4	65.4	55.0	66.0	58.2	64.0	56.7	64.6	55.0	64.8	58.2
5	65.0	58.4	64.8	58.2	64.2	58.1	65.4	56.5	64.8	57.8	64.3	56.1	63.1	57.1
6	65.4	56.5	65.2	58.4	65.5	55.8	65.6	56.0	64.6	57.0	65.1	56.0	65.3	56.6
Column Average	65.3	58.4	65.0	47.4	65.0	56.7	65.6	56.1	64.8	56.2	65.1	55.5	64.7	57.1

Grand Average UT: 65.1
LT: 56.5

UT = upper threshold
$ LT = lower threshold
*** Thresholds in db CSPL
+ Partial varied in relation to the other six partials.

Table 14

Treatment x Subjects ANOVA: Upper Threshold Data from Table 13

Source of Variation	Sum of Squares	Degrees of Freedom	Mean Square	F Ratio
Subjects	3.59	5	.72	
Partials	3.22	6	.54	2.18
Error (treatment x subject)	7.38	30	.24	
Total	14.19	35		

Table 15

Treatment x Subjects ANOVA: Lower Threshold Data from Table 13

Source of Variation	Sum of Squares	Degrees of Freedom	Mean Square	F Ratio
Subjects	14.75	5	2.95	
Partials	15.2	6	2.53	1.92
Error (treatment x subject)	39.52	30	1.32	
Total	69.47	35		

12). In no case did a subject respond "different" to two adjacent points on the method of limits arrangement outlined by Table 9. Therefore, it was impossible to determine a difference threshold for loudness related to changes in the amplitude spectrum of the sound defined by Equation 12, using the amplitude parameters of Table 9.

Discussion

The results of this investigation are not in agreement with those of Stumpf (quoted in Winckel, 1967, p. 113) that the amplitudes of single partials within a sound spectrum can be changed greatly before a distortion of the "tone color" is noticed. The results of this investigation also dispute his claim that an entire partial can be erased with no resultant change in timbre. Due to poor documentation of the Stumpf investigation, it is not possible to know what specific sound stimuli he used, nor the kinds of subjects (other than the fact that they had "untrained" ears).

This investigation's results show a clear ability of subjects to discriminate changes in timbre as the amplitude of a component was varied in relation to the rest of the spectrum. In fact, the results showed that there was no significant difference for results obtained with the varying of different partials. In this investigation, changing the amplitude of one component with redistribution among the other components results in a seven dimensional physical alteration, which is represented in the single decibel value of the varied partial by default. Stumpf's investigation did not reference itself to the sound's multidimensional physical structure. When a partial was increased or

decreased in amplitude, the other components of the spectra remained unaltered. It is not clear whether Stumpf investigated timbre changes or loudness changes, since he did not carefully control the power spectrum within each critical band, nor did he conduct an investigation to determine the effect of increasing or decreasing a single component of his sound on loudness, as was done in this investigation.

Researchers wishing to describe a change in tone color of an instrument as a function of reed hardness, ligature construction, bell dimensions, etc., should consider the results of this and other investigations. Many prior studies describe differences between physical components of a musical instrument in terms of changes in the resultant waveform produced by the instrument. Plomp and Steeneken (1969) have shown that two dissimilar waveforms may have similar timbres. The present study demonstrates that there is an area of subjective equality of timbre for a given waveform, and that two waveforms of similar spectral shape may not be just noticeably different from one another in timbre. Therefore, to simply describe the effect of changing some physical aspect of a musical instrument in terms of changes in the waveform is suspect. Whether such changes are just noticeably different from a standard must be first ascertained.

To the writer's knowledge this is the first study to systematically control loudness, pitch, and waveform in investigating timbre. The results of this study permit a new definition of timbre: Timbre is the sensation elicited by a tone when pitch, loudness, and phase spectra are held constant and amplitude spectrum is altered. This is not a negative definition, as were many of those found in the first chapter.

It relates the sensation of timbre to one physical aspect, amplitude spectrum.

Limitations

Two subjects in the present study claimed that they could "hear out" the higher individual components of the comparison stimuli in some cases. When asked to match a probe tone introduced by a sine oscillator to any component of the comparison stimuli, neither were able to do so, instead matching the frequency several hundred Hz above or below adjacent components. There was no control of the phase of the introduced probe tone, however. Whether the perceived subjective tones were due to combination tone effects is not clear. However, there is some experimental evidence to confirm the emphasis of the audibility of a partial in a setting such as that used for the present investigation. Plomp (1964) suggests that the slope of the excitation pattern that is not masked by a neighboring component contributes to the audibility of the partial, and that observers could distinguish the first five to eight harmonics of a complex tone (see Chapter 2). As a partial in the present investigation was increased in IL, the other partials were decreased proportionally; this increases the slope of the excitation pattern relative to the varying partial. In some cases, the subjects may have been correct in their ability to identify a partial, particularly at the extremes in physical dissimilarity between the standard and comparison stimulus. What clues this may provide a subject in detecting timbre changes is not clear; it is assumed that the conditions required to detect partials as separate entities rather than fused tones are beyond the values for the difference threshold for timbre.

Further investigation in this area is suggested. Also, the concept of fusion in complex sounds may be related to a subjective "compact-scattered" dimension; the relation of this to slope of power spectra also needs attention.

Several subjects also reported being disturbed by the abrupt onset of the stimuli used in this investigation. Many subjects felt that listening to the "center" of the stimulus aided them in making accurate decisions. The perceived abrupt onset is not predicted by the equation of the stimulus, since each component of the complex stimulus was in phase. Inaccuracies in timing during digital to anoly conversion are suspect.

In addition to the possibility that the individual components of the complex tone might be resolved by a subject and the annoyance of the abrupt onset of the signals, several other limitations or confounding factors can be identified. The resultant waveform from the tape was not exactly the same as that obtained by plotting points on a time vs. amplitude axis using radians for phase angle in Equation 12. The headphone response curves were linear within the frequency range of the stimuli used, but the tape recording equipment is suspect. This introduced bias was consistent throughout the experiment, as the same tape and tape recorder were used for the entire investigation. If real-time digital sound synthesis were employed, phase distortions due to tape to tape recorder transfer could be eliminated.

Other possible introduced biases have been reported elsewhere (Gescheider, 1976), and are inherent in any study using paired comparisons. Changing an observer's criterion over a series of trials

was minimized by rest periods and feedback from the computer display, but nevertheless could influence the results. Changes of criterion average over a series of trials, but cannot be completely eliminated from the present form of methodology. Fatigue is another factor that could influence the results. Again, frequent rest sessions assisted in this area, however the amount of sleep the subject received the night before the experiment, the amount of concentration required by the day's activities up to the time of the investigation, and other such factors cannot be eliminated. Again, such influences average over the series of trials conducted on two separate days.

Summary

Responses of six subjects to a series of standard and comparison randomly presented complex stimuli showed consistency enabling the determination of transition points where a difference in timbre was just discriminated between members of the stimulus pairs. Difference thresholds calculated using these transition points were not significantly different between partials used as treatment conditions in a treatment by subject analysis of variance. The mean difference threshold of 4.28 db CSPL was thus the average amount of energy that must be added or subtracted from a given partial of a seven component complex tone defined by Equation 12 (with redistribution of this energy among the other six partials) for a difference in timbre to be ascertained 50 percent of the time.

CHAPTER V

SUMMARY AND CONCLUSIONS

Summary

Evidence presented in Chapter 1 indicated that timbre had received considerably less attention in major works on psychoacoustics than the other primary attributes of tones. Indeed, the term itself often was defined differently in various scholarly sources. The narrow definition of timbre, rather than the "catch-all" definition used by many investigators, was accepted as applicable to the present study.

Although the basic questions guiding investigations in psychoacoustics had been answered for loudness and pitch of both sinusoidal and complex tones, these questions remained unanswered for timbre. Lack of research was attributed to the multidimensional nature of timbre, and the fact that generation of complex tones with specific parameters only recently has been aided with the use of digital computers.

The present study sought to investigate with timbre some of the basic questions central to psychophysics: What is the smallest amount of change in physical energy that the (biological) system can react to? How does the system react to variations in the amount of energy received? To answer these questions the physical nature of the complex tone was analyzed. The power spectrum was reported to be

timbre's primary physical correlate (Plomp, 1971). The research questions which arose were: 1. What is an individual's sensitivity to timbre changes as a function of the intensity change of one of the partials of a complex sound? 2. Is this sensitivity to timbre differences the same regardless of which partial is varied? 3. How do loudness and timbre relate to power spectrum? To answer these questions, the difference threshold was determined for timbre as related to the power spectra of complex sounds. The null hypothesis was: There will not be a significant difference among means for difference thresholds obtained by varying each of the seven components of a complex sound.

Literature was reviewed in three distinct areas: 1. Difference thresholds; 2. loudness, pitch, and phase aspects of multicomponent stimuli; 3. timbre. The method of limits paradigm was the most promising procedure for determining the difference threshold because the stimuli did not necessarily have to lie along a sensory continuum.

Studies with multicomponent stimuli indicate that the resolving power of the ear is about 1/3 octave (Zwicker, 1954). Loudness of complex tones is approximately equal to the sum of the loudness contributions of each of the critical bands (Stevens, 1961, 1972; Zwicker, Flottrop, Stevens, 1957). The pitch of a complex tone is attributable to a variety of physical aspects; the psychological results are not altogether clear.

The study of timbre as related to the amplitude spectra of complex sounds was initiated by Seebeck (1849), and later expanded

upon by Helmholtz (1877). Early attention to timbre perception focused on vowel-like sounds, which were related to peaks in the amplitude spectrum known as formants (Hermann, 1890). Experimentation in this area lead to the first claim for a difference threshold for timbre, related to the absolute frequency of the formants (Winckel, 1967).

Phase effects on timbre apparently are greatest as the phase diverges from 0° (Raiford and Schubert, 1971), and the greatest dissimilarity in timbre is between tones comprised of all sine terms and tones with alternating sine and cosine terms (Plomp and Steeneken, 1969). The effect of phase on timbre is smaller than the effect of varying the amplitude spectrum (Plomp and Steeneken, 1969; Plomp, 1970).

Some experimentation with amplitude spectra as related to timbre claimed that an entire partial could be erased without being perceptible, as long as the partial was not a formant (Stumpf, quoted in Winckel, 1967). Plomp (1970) found a high correlation between the physical space of a complex sound (as represented by the power spectrum) and the psychological space for timbre. Grey (1976) identified three dimensions that were most important in describing the timbre elicited by complex tones: power spectrum, attack, and fine structure fluctuations.

The review of the literature revealed that a study in the timbre difference threshold related to power spectrum had not been reported. Since there was no model upon which to base such an investigation, a series of pilot studies were conducted to determine the specifics of the values to be given to the amplitude spectrum and the specific methodology to employ.

Digital sound synthesis was determined to be the most economical means of producing complex sounds with precise control of the parameters. The investigator used Music V sound generation software, developed by Max Mathews of Bell Laboratories (1969), and implemented at The University of Kansas by Roy Campbell, to generate the complex sound stimuli used in all facets of the investigation. The first pilot generated a 500 Hz sine wave with a relative amplitude of 50 db. With a sampling rate of 30,000 samples/sec, the output was distorted. However, at 17,500 samples/sec sampling rate the output, when passed into an oscilloscope, had the appearance of a sine wave, although there were peaks representing distortion along the curvature of the function. A modification of the oscillator software involving replacing the truncating function with an interpolating function eliminated much of the distortion.

The methodology employed to determine the difference threshold was the method of limits because it did not require real-time sound synthesis (which was beyond the scope of the Music V program and its associated hardware) and because it could be adapted for responses from a 2AFC procedure.

The standard stimulus was a complex sound consisting of seven sine components with a fundamental frequency of 500 Hz; the components were in an integral multiple relation to the fundamental, of equal amplitude, and were in phase. Each was within a separate critical band. The stimulus was within the 8,750 Hz cut-off of computer synthesis based on a 17,500 Hz sampling rate.

In the first pilot study, increments of .1 db above and below the standard were used for varying the intensity of the 7th partial. Energy gained or lost from the 7th partial was redistributed among the other six partials; the total intensity was 49.95 db IL. The standard stimulus was presented first and followed by a comparison stimulus; each tone lasted 2 sec. Three seconds were allowed for the subject to decide whether each pair contained tones of identical or different timbre.

The results of one trained subject indicated that no differences between the pairs were heard. When .5 db increments were used the results were the same.

The second pilot study varied the fourth partial in relation to the other six in increments of 1.569×10^{-9} w/m^2. The amount of time between comparison and standard stimuli was shortened to 1.5 sec. Comparison stimuli were presented from the lowest energy in partial four to the most. There were nine increments above and below the standard stimulus.

Results of one trained subject indicated a "different" response at 34.99 db (relative amplitude) each time for three trials. No upper threshold was discernable as all responses with added energy in the 4th partial were "same."

The third pilot sutdy used a larger 6.334×10^{-9} w/m^2 increment with the same procedure as in the second pilot. Results of one trained subject for three trials of the 19 stimulus pairs were consistent enough to establish a difference threshold of 4.775 db CSPL.

Pilot study 4, in which computer amplitude units were used to represent linear amplitude measurements, failed in digital sound processing.

Pilot study 5 increased the resolution of the experimental paradigm by taking the area of subjective equality found in the third study and deleting part of it. The presentation order of comparison and standard stimuli was completely randomized to minimize errors due to expectation.

Results of two subjects indicated that absolutely consistent responses did not occur; extraneous responses among a series of consistent responses were evident.

A criterion for determining the transition point was established; the difference threshold was the point that responses changed from one state to another with at least two responses in the new direction.

Pilot studies 6, 7, and 8 were conducted with various IL increments designed to increase the resolution of the paradigm. The final investigation was formulated from the results of the first eight pilot studies.

Six subjects were solicited from music camp participants and music students at The University of Kansas: four subjects were music campers, one was an undergraduate music student, and one was a graduate music student. Each subject participated in one training and seven measurement sessions over two days.

The training session consisted of an explanation of the investigation conducted by interaction with a program on a Radio

Shack TRS-80 microcomputer. This computer also recorded the subjects "same" and "different" responses.

The modified method of limits methodology was used, with complete randomization of standard and comparison stimuli. The standard stimulus was the complex of simultaneous sinusoids defined by

$$\underline{p}(\underline{t}) = \sum_{n=1}^{m=7} \underline{a}_n \sin (2\Pi \underline{n} * 500 * \underline{t} + \Phi).$$

The comparison stimuli had energy added or subtracted from a given partial of the standard and redistributed among the other six partials according to the values in Table 9. There were three separate random orders for each of the seven partials used as levels of the independent variable. Each member of the stimulus pair was presented for two seconds, with one second of silence between members of the pair. Four seconds of silence were allowed between pairs of stimuli for the subject's response.

The stimuli were presented monaurally to the left ear at 70 db SPL as measured with a Telephonics model 49 headphone coupled with a Bruel and Kjaer model 5152 acoustic coupler and model 4144 microphone model 1613 octave filter. The sounds produced by digital-to-analog conversion at a 17,500 samples/sec sampling rate were recorded on Maxell UD recording tape and played through a MacIntosh model C-26 preamplifier. The measurement sessions were conducted in the Psychology of Music Laboratory at The University of Kansas.

For each of the six measurement sessions, the subject responded to seven sets of 19 stimulus pairs--one set for each partial varied in the seven-component complex tone. Each partial varied in relation to the rest of the complex produced six data sets per subject.

Three additional subjects as well as those participating in the timbre investigation were asked to respond to a random tape of stimuli varying the third partial, only this time they were instructed to listen for loudness differences between pairs.

The responses to the randomly ordered stimuli were sequentially ordered by a computer program. The transitional points were the intensities of the partial varied in relation to the others (in db CSPL) where the "same" responses changed to "different," with at least two responses in the new direction. The upper and lower thresholds were the midpoints between these transitional points and the next stimulus value in each direction. The difference thresholds were calculated using Equations 13 and 14, and were summarized in Table 10.

A treatment by subject analysis of variance with partial varied as the treatment conditon, indicated no significant difference between column means, thereby supporting the hypothesis of no significant difference in difference thresholds obtained by varying each of the seven components of a complex sound as defined by Equation 12. No significant difference was found for upper or lower threshold data subjected to a similar treatment by subject ANOVA. Therefore, an average difference threshold for timbre for the standard stimulus was calculated, and found to be 4.28 db CSPL.

No subject was able to give consistent responses for loudness changes between stimuli used in the timbre investigation varying the third partial. A difference threshold for loudness could not be calculated using the amplitude values of Table 9.

Conclusions

Using a standard stimulus of seven sine components which are in integral multiple relation to one another and have equal amplitudes, it is possible to specify the amount of energy that must be added to or subtracted from one of the partials and redistributed among the others before a change in timbre can be perceived. The results for six subjects yielded a difference threshold for timbre for each of the seven partials varied. There was no significant difference among the difference thresholds obtained for different partials.

It can be concluded that, for the standard stimulus defined by Equation 12, a larger power spectrum change is required to induce a change of loudness than to induce a change of timbre because no systematic responses to loudness changes were produced by subjects responding to stimuli with amplitude values specified by Table 9, and such systematic responses were evidenced for timbre. Because loudness investigations only were conducted with the third partial varied, these results are limited to varying that partial; additional study would be required to generalize beyond this.

Any study which analyzes the complex waveform of a sound cannot assume that waveform changes yield timbre changes. Experimentation must be conducted to ascertain whether a change in timbre is induced by varying changes in the power spectrum.

For the stimulus used in this study, it appears that each critical band contributes equally to the overall timbre (as implied by Plomp (1971). However, if there were more than one component in each critical band, the results of this study would not generalize. This

suggests an area for further study.

Suggestions for Further Study

A natural follow-up investigation would replicate this study with a greater number of subjects. A major improvement of the present design would be to use real-time digital sound synthesis. Not only would different experimental paradigms be possible (such as the method of limits tracking technique developed by Von Bekesy), but influences due to tape recorded sound could be eliminated.

This study only addresses a single standard stimulus, one that is not readily found in nature. Much of psychoacoustics has used standardized waveforms and frequencies (evidenced in the use of sine tones and fixed frequency standards for definitions), so the use of such a standard is not without precedent. Nevertheless, work with other standard stimuli is desirous. A standard stimulus with a given slope could be used, rather than the stimulus with equal amplitude components used in this study. In such an investigation, the slope, expressed in db/octave, would be the dependent variable. With real-time sound synthesis, a procedure similar to Fant's (1958) might be used with amplitude spectrum, rather than formant frequency, varied by a digitized x-y potentiometer.

This study also should be replicated with more than one component of the sound complex systematically varied. By using a fundamental of 1,000 Hz, the varying of components lying within the same critical band could be studied, possibly answering the question: What influences the timbre of a complex sound the most, slope of the

power spectrum between critical bands or within a critical band, or does timbre depend most on the overall shape of the power spectrum, without reference to critical bandwidth?

Although timbre changes were induced by varying different partials of a complex by approximately the same amount, it is not clear whether changing partial 1 by 5 db SPL creates the same timbre as changing partial 3 by 5 db SPL (with redistribution among the other partials). This question can be resolved by using the method of triadic comparisons to develop a dissimilarity matrix for all possible pairs of tones used in this investigation.

In all areas of psychoacoustical research, much more must be done with studying perception of stimuli in musical context. Questions to be answered include: Are tones with a given frequency in ascending passages heard as having the same pitch as tones with the same frequency in descending passages? What are the difference thresholds for timbre, pitch, and loudness for tones in musical contexts? Of course, duration will be another aspect to consider. In a musical passage, is it possible to place a tone quality recognized by authorities as "poor" within a fast moving passage, and have the passage end with a whole note of "good" tone quality, and have the entire event perceived as "good"? Rating scales coupled with systematic manipulation of timbre and durational elements could yield interesting, and practical, results.

One must not be afraid to research within a narrow context. Too often "universal truths" are pursued. Research in a few areas may lead toward larger understandings. The loudness of the sine wave was not measured by the same procedure as the loudness of a complex tone, but understanding the first led (much later) to the latter.

Eventually, timbre research may enable quantification of timbre in a meaningful way, taking into account aspects of spectral fluctuations in the fine structure and onset characteristics. If a single index of timbre can be developed, it will be possible to quantify the comparison of two similar tone qualities. By use of rating scales and judgement procedures, it would be possible to determine, perhaps by real-time analysis, how much better a teacher's tone quality is from a student's, for example. There might be hesitancy in this area because of the need for a standard, but designated pitches and relative loudnesses have been used for centuries (e.g., staff notation and markings such as mf and pp). Composers of the 20th century are aware that timbre from like instruments can vary. Witness the use of "brassy," "strident," "mellow," and the like in recent scores. In the future, it may be possible to relate these semantic descriptors to specific physical aspects. A relation has been shown (Von Bismarck, 1974). Researchers now must find exactly what the relation is.

Timbre is a complex, multidimensional phenomenon. The difficulty of its study is a challange for musicians and psychoacousticians. Its complexity does not justify ignorance, even if the first steps are very small.

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APPENDIX A

Music V Program Listing

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CPASS1          PASS 1 MAIN PROGRAM                                  00000010
CPASS1             *** MUSIC V ***                                   00000020
      DIMENSIONP(100),D(2000),IP(10)                                 00000030
      COMMONIP,P,D                                                   00000040
C     INIALIZE PROGRAM                                               00000050
C          NOMINAL SAMPLING RATE.                                    00000060
      CALL READ0                                                     00000070
      D(4)=10000.0                                                   00000080
      REWIND10                                                       00000090
C     MAIN DATA PROCESSING LOOP.                                     00000100
 100  CALLREAD1                                                      00000110
      I1=P(1)                                                        00000120
      IF(I1)101,101,102                                              00000130
 102  IF(I1-12)103,103,101                                           00000140
 101  CALLERROR(10)                                                  00000150
      GOTO100                                                        00000160
 103  GOTO(1,1,1,1,5,6,7,1,9,1,1,12),I1                              00000170
 1    CALLWRITE1(10)                                                 00000180
      GOTO100                                                        00000190
 6    CALLWRITE1(10)                                                 00000200
      PRINT111                                                       00000210
 111  FORMAT(13HEND OF PASS I)                                       00000220
      IF(IP(2).EQ.1) CALL HARVEY                                     00000230
      STOP                                                           00000240
 7    I2=P(3)                                                        00000250
      I3=I2+IP(1)-4                                                  00000260
      DO104I4=I2,I3                                                  00000270
      I5=I4-I2+4                                                     00000280
 104  D(I4)=P(I5)                                                    00000290
      GOTO100                                                        00000300
 9    I6=P(3)                                                        00000310
      IF(I6)106,106,105                                              00000320
 105  IF(I6-5)107,107,106                                            00000330
 106  CALLERROR(11)                                                  00000340
      GOTO100                                                        00000350
 5    PRINT110                                                       00000360
 110  FORMAT(21HEND OF SECTION PASS I)                               00000370
      GOTO1                                                          00000380
 12   CALLWRITE1(10)                                                 00000390
      GOTO7                                                          00000400
 107  GOTO(21,22,23,24,25),I6                                        00000410
 21   CALLPLF1                                                       00000420
      GOTO100                                                        00000430
 22   CALLPLF2                                                       00000440
      GOTO100                                                        00000450
 23   CALLPLF3                                                       00000460
      GOTO100                                                        00000470
 24   CALLPLF4                                                       00000480
      GOTO100                                                        00000490
 25   CALLPLF5                                                       00000500
      GOTO100                                                        00000510
      END                                                            00000520
```

```
CWRIT1          PASS 1 DATA-WRITING ROUTINE                    00000530
C                 *** MUSIC V ***                              00000540
      SUBROUTINEWRITE1(N)                                      00000550
      DIMENSIONIP(10),P(100)                                   00000560
      COMMONIP,P                                               00000570
      K=IP(1)                                                  00000580
      WRITE(N )K,(P(J),J=1,K)                                  00000590
      RETURN                                                   00000600
      END                                                      00000610
```

```
      SUBROUTINE PLF1                                                   00000620
      COMMON IP,P,D                                                     00000630
      DIMENSION IP(10),P(100),D(2000)                                   00000640
      NS=P(4)                                                           00000650
      NE=P(5)                                                           00000660
      TS=P(6)                                                           00000670
      DS=P(7)                                                           00000680
      FS=P(8)                                                           00000690
      IP(1)=6                                                           00000700
      P(1)=1.0                                                          00000710
      P(3)=P(9)                                                         00000720
      DO 100 I=NS,NE,10                                                 00000730
      P(2)=TS+DS*D(I)                                                   00000740
      P(4)=DS*D(I+1)                                                    00000750
      P(5)=D(I+2)                                                       00000760
      P(6)=(2.0**(D(I+3)+FS))*262.0                                     00000770
      CALL WRITE1(10)                                                   00000780
  100 CONTINUE                                                          00000790
      RETURN                                                            00000800
      END                                                               00000810
```

```
      SUBROUTINE PLF2                                                   00000820
      COMMON IP,P,D                                                     00000830
      DIMENSION IP(10),P(100),D(2000)                                   00000840
      NB1=P(4)                                                          00000850
      NE1=P(5)                                                          00000860
      NB2=P(6)                                                          00000870
      NE2=P(7)                                                          00000880
      TS=P(8)                                                           00000890
      IP(1)=6                                                           00000900
      P(1)=1.0                                                          00000910
      P(3)=P(9)                                                         00000920
      DO 101 I=NB1,NE1,10                                               00000930
      START=TS+D(I)                                                     00000940
      DS=D(I+1)/(D(NE2)+D(NE2+1))                                       00000950
      DO 100 J=NB2,NE2,10                                               00000960
      P(2)=START+DS*D(J)                                                00000970
      P(4)=DS*D(J+1)                                                    00000980
      P(5)=D(J+2)+D(I+2)                                                00000990
      P(6)= 2.D**(D(J+3)+D(I+3))*262.0                                  00001000
      CALL WRITE1(10)                                                   00001010
  100 CONTINUE                                                          00001020
  101 CONTINUE                                                          00001030
      RETURN                                                            00001040
      END                                                               00001050
```

```
      SUBROUTINE PLF3                                                   00001060
      COMMON IP,P,D                                                     00001070
      DIMENSION IP(10),P(100),D(2000)                                   00001080
      TS=P(4)                                                           00001090
      END=P(5)                                                          00001100
      NA=P(6)                                                           00001110
      NP=P(7)                                                           00001120
      NDR=P(8)                                                          00001130
      NDF=P(9)                                                          00001140
      P(1)=1.0                                                          00001150
      P(3)=P(10)                                                        00001160
      IP(1)=6                                                           00001170
      T=0.0                                                             00001180
  100 DR=CON(D,NDR,T)                                                   00001190
      IF(T+DR-END) 101,101,104                                          00001200
  101 P(2)=T+TS                                                         00001210
      P(4)=DR*CON(D,NDF,T)                                              00001220
      IF(P(4)) 103,103,102                                              00001230
  102 P(5)=CON(D,NA,T)                                                  00001240
      P(6)=CON(D,NP,T)                                                  00001250
      CALL WRITE1(10)                                                   00001260
  103 T=T+DR                                                            00001270
      GO TO 100                                                         00001280
  104 RETURN                                                            00001290
      END                                                               00001300
```

```
      SUBROUTINE PLF4                                                   00001310
      COMMON IP,P,D                                                     00001320
      DIMENSION IP(10),P(100),D(2000)                                   00001330
      TS=P(4)                                                           00001340
      END=P(5)                                                          00001350
      NA=P(6)                                                           00001360
      NP=P(7)                                                           00001370
      I=NA                                                              00001380
      IP(1)=5                                                           00001390
      P(1)=4.0                                                          00001400
      P(3)=3.0                                                          00001410
  100 P(4)=10.0**(D(I+1)/20.0)                                          00001420
      P(5)=(10.0**(D(I+3)/20.0)-P(4))/((D(I+2)                          00001430
     2-D(I))*D(4))                                                      00001440
      P(2)=TS+D(I)                                                      00001450
      CALL WRITE1(10)                                                   00001460
      IF(D(I+2)-END) 101,102,102                                        00001470
  101 I=I+2                                                             00001480
      GO TO 100                                                         00001490
  102 I=NP                                                              00001500
      P(3)=5.0                                                          00001510
  103 P(4)=2.0**D(I+1)                                                  00001520
      P(5)= (2.0**D(I+3)-P(4))/((D(I+2)-D(I))*D(4))                     00001530
      P(2)=TS+D(I)                                                      00001540
      CALL WRITE1(10)                                                   00001550
      IF(D(I+2)-END) 104,105,105                                        00001560
  104 I=I+2                                                             00001570
      GO TO 103                                                         00001580
  105 IP(1)=4                                                           00001590
      P(1)=1.0                                                          00001600
      P(2)=TS                                                           00001610
      P(3)=1.0                                                          00001620
      P(4)=END                                                          00001630
      CALL WRITE1(10)                                                   00001640
      P(3)=2.0                                                          00001650
      CALL WRITE1(10)                                                   00001660
      RETURN                                                            00001670
      END                                                               00001680
```

```
      CREAD1                INTERPRETATIVE READING ROUTINE                  00001690
      C                    ****MUSIC V****                                  00001700
            SUBROUTINEREAD1                                                 00001710
            COMMONIP,P,D                                                    00001720
            DIMENSIONP(100),D(2000),IP(10)                                  00001730
            DIMENSIONCARD(129),ICAR(128),IBCD(300),LOP(3),3)                00001740
            DIMENSIONBCD(300)                                               00001750
            DIMENSIONIBC(12),IVT(4)                                         00001760
            EQUIVALENCE(CARD,ICAR)                                          00001770
            EQUIVALENCE(BCD,IBCD)                                           00001780
            DATANOPS,NBC,NC/26,3,72/                                        00001790
            DATAIBEC,ISTAR/6H00000.,6H00000*/                               00001800
            DATA(IBC(I),I=1,4)/6H00000;,6H00000 ,6H00000,,6H00000-/         00001810
            DATA(IVT(I),I=1,4)/6H00000P,6H00000F,6H00000B,6H00000V/         00001820
            DATA((LOP(I,J),J=1,3),I=1,26)/6H00000N,6H000000,6H000001,6H000001,  00001830
           16H00000N,6H00000S,6H000006,6H00000E,6H00000N,6H00000S,6H00000V,  00001840
           26H00000D3,6H00000S,6H00000E,6H00000C,6H000001,6H00000E,6H00000R,  00001850
           36H00000S,6H00000V,6H000001,6H00000S,6H00000V,6H000002,6H00000P,  00001860
           46H00000L,6H00000F,6H00000P,6H00000L,6H00000S,6H00000S,6H00000I,  00001870
           56H000003,6H00000S,6H000001,6H00000A,6H00000C,6H000000,6H00000M,  00001880
           66H00000E,6H00000N,6H00000D,6H000000,6H00000U,6H00000T,6H000000,  00001890
           76H00000S,6H00000C,6H00000A,6H00000D,6H000002,6H00000R,6H00000A,  00001900
           86H00000N,6H00000E,6H00000N,6H00000V,6H00000S,6H00000T,6H00000R,  00001910
           96H00000A,6H00000D,6H000003,6H00000A,6H00000D,6H000004,6H00000M,  00001920
           16H00000L,6H00000T,6H00000F,6H00000L,6H000001,6H00000R,6H00000A,  00001930
           26H00000M,6H00000S,6H00000E,6H00000T/                            00001940
      C     TO SCAN INPUT DATA TO ;, ORGANIZE FIELDS AND PRINT              00001950
            IF(END+SMAB-1.)10,10,90                                         00001960
      10    IBK=2                                                           00001970
            END=0.                                                          00001980
            ERR=0.                                                          00001990
            NUMU=0                                                          00002000
            ISEMI=1                                                         00002010
            L=3                                                             00002020
            J=0                                                             00002030
      11    I=I+1                                                           00002040
            IF(I.GT.NC)GOTO15                                               00002050
            IF(J.EQ.299)GOTO21                                              00002060
            DO13N=1,NBC                                                     00002070
            IF(ICAR(I)-IBC(N))13,12,13                                      00002080
      12    GOTO(20,16,18),N                                                00002090
      13    CONTINUE                                                        00002100
            J=J+1                                                           00002110
            IBCD(J)=ICAR(I)                                                 00002120
            IBK=1                                                           00002130
            GOTO11                                                          00002140
      14    IBK=N                                                           00002150
            GOTO11                                                          00002160
      15    READ1,(CARD(I),I=1,NC)                                          00002170
      1     FORMAT(128A1)                                                   00002180
            PRINT 2,(CARD(I),I=1,NC)                                        00002190
      2     FORMAT(1H 128A1)                                                00002200
```

```
      IF(IFOF.EQ.1)GOTO95                                          00002210
      CALLMOVR(CARD,MC)                                            00002220
      I=0                                                          00002230
      GOTO11                                                       00002240
16    GOTO(17,11,11),IBK                                           00002250
17    IBK=M                                                        00002260
      J=J+1                                                        00002270
      IBCD(J)=IBC(2)                                               00002280
      GOTO(11,21),ISEM1                                            00002290
18    GOTO(17,14,19),IBK                                           00002300
19    J=J+1                                                        00002310
      IBCD(J)=0                                                    00002320
      GOTO17                                                       00002330
20    ISEM1=2                                                      00002340
      GOTO(17,21,19),IBK                                           00002350
21    J=J+1                                                        00002360
      IBCD(J)=IBC(1)                                               00002370
C     TO SCAN FOR OP CODE                                          00002380
      DO24N=1,NOPS                                                 00002390
      M=N                                                          00002400
      DO23K=1,3                                                    00002410
      IF(IBCD(K)-LOP(N,K))24,23,24                                 00002420
23    CONTINUE                                                     00002430
      GOTO26                                                       00002440
24    CONTINUE                                                     00002450
      GOTO40                                                       00002460
26    NP=1                                                         00002470
27    L=L+1                                                        00002480
      IF(IBCD(L)-IBC(2))27,29,27                                   00002490
29    GOTO(100,200,300,400,500,600,700,800,900,1000,1100,1200,1300,217,  00002500
     1201,202,203,204,205,206,207,208,209,210,211,212),M           00002510
C     OP CODE 1 TO PLAY NOTE                                       00002520
100   P(1)=1.                                                      00002530
      GOTO30                                                       00002540
C     OP CODE 2 TO DEFINE INSTRUMENT                               00002550
200   P(1)=2.                                                      00002560
      IDEF=1                                                       00002570
      N1=1                                                         00002580
      GOTO70                                                       00002590
2000  P(2)=XN                                                      00002600
      N1=2                                                         00002610
      GOTO70                                                       00002620
2001  P(3)=XN                                                      00002630
      IP(1)=3                                                      00002640
      GOTO50                                                       00002650
C     OUT BOX                                                      00002660
201   P(3)=101.                                                    00002670
      NPW=2                                                        00002680
      IF(STER)220,220,2011                                         00002690
2011  SNAB=1.                                                      00002700
      STER=0.                                                      00002710
      GOTO220                                                      00002720
```

```
105      C      OSCILLATOR                                          00002730
106       202   P(3)=102.                                           00002740
107             NPW=5                                               00002750
108             GOTO220                                             00002760
109      C      ADD 2                                               00002770
110       203   P(3)=103.                                           00002780
111             NPW=3                                               00002790
112             GOTO220                                             00002800
113      C      RANDOM AND INTERPOLATE                              00002810
114       204   P(3)=104.                                           00002820
115             NPW=6                                               00002830
116             GOTO220                                             00002840
117      C      LINEAR ENVELOPE GENERATOR                           00002850
118       205   P(3)=105.                                           00002860
119             NPW=7                                               00002870
120             GOTO220                                             00002880
121      C      STEREO OUT BOX                                      00002890
122       206   P(3)=106.                                           00002900
123             NPW=3                                               00002910
124             IF(STER)220,2061,220                                00002920
125       2061  SNAB=1.                                             00002930
126             STER=1.                                             00002940
127             GOTO220                                             00002950
128      C      THREE INPUT ADDER                                   00002960
129       207   P(3)=107.                                           00002970
130             NPW=4                                               00002980
131             GOTO220                                             00002990
132      C      FOUR INPUT ADDER                                    00003000
133       208   P(3)=108.                                           00003010
134             NPW=5                                               00003020
135             GOTO220                                             00003030
136      C      MULTIPLIER                                          00003040
137       209   P(3)=109.                                           00003050
138             NPW=3                                               00003060
139             GOTO220                                             00003070
140      C      FILTER                                              00003080
141       210   P(3)=112.                                           00003090
142             NPW=4                                               00003100
143             GOTO220                                             00003110
144      C      RANDOM AND HOLD                                     00003120
145       211   P(3)=111.                                           00003130
146             NPW=5                                               00003140
147             GOTO220                                             00003150
148      C      SET NEW FUNCTION                                    00003160
149       212   P(3)=110.                                           00003170
150             NPW=1                                               00003180
151             GOTO220                                             00003190
152      C      END OF INSTRUMENT                                   00003200
153       217   IP(1)=2                                             00003210
154             IDEF=0                                              00003220
155             END=1.                                              00003230
156             GOTO50                                              00003240
```

```
157      C      UNNAMED UNIT   -  NUMERICAL NAME ASSUMED                    00003250
158       218   N1=8                                                       00003260
159             NUMU=1                                                     00003270
160             L=0                                                        00003280
161             GOTO70                                                     00003290
162       219   N=XN+14.                                                   00003300
163             IF(XN.LT.11.)GOTO29                                        00003310
164             P(3)=XN                                                    00003320
165      C      TO INTERPRET VARS IN UNIT DEFS                             00003330
166       220   NP=3                                                       00003340
167       221   IF(IBCD(L+1)-IBC(1))222,240,222                            00003350
168       222   NP=NP+1                                                    00003360
169             L=L+1                                                      00003370
170             DO223N=1,4                                                 00003380
171             IF(IBCD(L)-IVT(N))223,225,223                              00003390
172       223   CONTINUE                                                   00003400
173       224   L=L+1                                                      00003410
174             IF(IBCD(L).EQ.IBC(2))GOTO46                                00003420
175             GOTO224                                                    00003430
176       225   GOTO(231,232,233,234),N                                    00003440
177      C      P TYPE                                                     00003450
178       231   N1=3                                                       00003460
179             GOTO70                                                     00003470
180       2311  P(NP)=XN                                                   00003480
181             GOTO221                                                    00003490
182      C      F TYPE                                                     00003500
183       232   N1=4                                                       00003510
184             GOTO70                                                     00003520
185       2321  P(NP)=-(XN+100.)                                           00003530
186             GOTO221                                                    00003540
187      C      B TYPE                                                     00003550
188       233   N1=5                                                       00003560
189             GOTO70                                                     00003570
190       2331  P(NP)=-XN                                                  00003580
191             GOTO221                                                    00003590
192      C      V TYPE                                                     00003600
193       234   N1=6                                                       00003610
194             GOTO70                                                     00003620
195       2341  P(NP)=XN+100.                                              00003630
196             GOTO221                                                    00003640
197       240   IF(NUMU.EQ.1)GOTO242                                       00003650
198       241   IF(NPW+3-NP)42,242,42                                      00003660
199       242   IP(1)=NP                                                   00003670
200             GOTO50                                                     00003680
201      C      OP CODE 3 - TO GENERATE FUNCTION                           00003690
202       300   P(1)=3.                                                    00003700
203             GOTO30                                                     00003710
204      C      OP CODE 4 - TO SET PARAM 3RD PASS                          00003720
205       400   P(1)=4.                                                    00003730
206             GOTO30                                                     00003740
207      C      OP CODE 5 TO END SEC                                       00003750
208       500   P(1)=5.                                                    00003760
```

```
      GOTO30                                                            00003770
C     OP CODE 6 TO TERMINATE PIECE                                      00003780
 600  P(1)=6.                                                           00003790
      GOTO30                                                            00003800
C     OP CODE 7 TO SET PARAM 1ST PASS                                   00003810
 700  P(1)=7.                                                           00003820
      GOTO30                                                            00003830
C     OP CODE 8 TO SET PARAM 2ND PASS                                   00003840
 800  P(1)=8.                                                           00003850
      GOTO30                                                            00003860
C     OP CODE 9 TO EXECUTE SUB 1ST PASS                                 00003870
 900  P(1)=9.                                                           00003880
      GOTO30                                                            00003890
C     OP CODE 10 TO EXECUTE SUB 2ND PASS                                00003900
 1000 P(1)=10.                                                          00003910
      GOTO30                                                            00003920
C     OP CODE 11 TO SET INTEGER 3RD PASS                                00003930
 1100 P(1)=11.                                                          00003940
      GOTO30                                                            00003950
C     OP CODE 12 TO SET INTEGER ALL PASSES                              00003960
 1200 P(1)=12.                                                          00003970
      GOTO30                                                            00003980
C     OP CODE 13 FOR COMMENTS                                           00003990
 1300 IF(IBCD(L)-IBC(1))1301,10,1301                                    00004000
 1301 L=L+1                                                             00004010
      GOTO1300                                                          00004020
C     TO STORE PFIELDS                                                  00004030
 30   IF(IDEF)32,32,43                                                  00004040
 32   IF(IBCD(L+1)-IBC(1))33,34,33                                      00004050
 33   NP=NP+1                                                           00004060
      N1=?                                                              00004070
      GOTO70                                                            00004080
 331  P(NP)=XN                                                          00004090
      GOTO32                                                            00004100
 34   IP(1)=NP                                                          00004110
      IF(NP-1)47,47,50                                                  00004120
C     ERRORS                                                            00004130
 40   IF(IDEF)41,41,218                                                 00004140
 41   L=L+1                                                             00004150
      IF(IBCD(L).NE.IBC(1))GOTO41                                       00004160
      PRINT3                                                            00004170
 3    FORMAT(26H     OP CODE NOT UNDERSTOOD)                            00004180
      GOTO49                                                            00004190
 42   PRINT4                                                            00004200
 4    FORMAT(44H     UNIT CONTAINS WRONG NUMBER OF PARAMETERS)          00004210
      GOTO49                                                            00004220
 43   PRINT5                                                            00004230
 5    FORMAT(36H     INSTRUMENT DEFINITION INCOMPLETE)                  00004240
      ERR=1.                                                            00004250
      IDEF=0                                                            00004260
      GOTO32                                                            00004270
 44   PRINT6                                                            00004280
```

```
261      6    FORMAT(25H     ERROR IN NUMERIC DATA)                          00004290
262           ERR=1.                                                         00004300
263           IF(NUMU.EQ.1)GOTO45                                            00004310
264      45   PRINT7                                                         00004320
265      7    FORMAT(46H+                          FOR UNIT DESIGNATION)     00004330
266           P(3)=0.                                                        00004340
267           GOTO220                                                        00004350
268      46   PRINT8                                                         00004360
269      8    FORMAT(40H     IMPROPER VARIABLE IN UNIT DEFINITION)           00004370
270           ERR=1.                                                         00004380
271           GOTO221                                                        00004390
272      47   PRINT9                                                         00004400
273      9    FORMAT(24H     STATEMENT INCOMPLETE)                           00004410
274      49   IP(2)=1                                                        00004420
275           GOTO10                                                         00004430
276      50   IF(ERR.EQ.1.)GOTO49                                            00004440
277           RETURN                                                         00004450
278      C    CONVERSION OF NUMERIC FIELD TO FLOATING POINT                  00004460
279      70   SGN=1.                                                         00004470
280           IF(IBCD(L+1).NE.IBC(4))GOTO79                                  00004480
281           SGN=-1.                                                        00004490
282           L=L+1                                                          00004500
283      79   L1=L+1                                                         00004510
284           LD=L1                                                          00004520
285           XN=0.                                                          00004530
286      71   L=L+1                                                          00004540
287           IF(IBCD(L)-IBC(2))72,77,72                                     00004550
288      72   IF(IBCD(L).LT.10)GOTO71                                        00004560
289           IF(IBCD(L)-IDEC)74,71,74                                       00004570
290      74   IF(IBCD(L)-ISTAR)76,71,76                                      00004580
291      76   L=L+1                                                          00004590
292           IF(IBCD(L).EQ.IBC(2))GOTO44                                    00004600
293           GOTO76                                                         00004610
294      77   IF(IBCD(L1)-ISTAR)80,78,80                                     00004620
295      78   XN=P(NP)                                                       00004630
296           GOTO89                                                         00004640
297      80   DO81LL=L1,L                                                    00004650
298           LD=LL                                                          00004660
299           IF(IBCD(LL)-IDEC)81,82,81                                      00004670
300      81   CONTINUE                                                       00004680
301      82   IEX=0                                                          00004690
302           LA=L1                                                          00004700
303           LB=LD-1                                                        00004710
304           IF(LD-L1)86,86,83                                              00004720
305      83   IEX=LD-LA                                                      00004730
306      84   DO85LL=LA,LB                                                   00004740
307           IEX=IEX-1                                                      00004750
308           XI=IBCD(LL)                                                    00004760
309      85   XN=XN+XI*10.**IEX                                              00004770
310      86   IF(L-LB-2)88,88,87                                             00004780
311      87   LA=LD+1                                                        00004790
312           LB=L-1                                                         00004800
```

```
      GOTO84                                                    00004810
 86   XN=XN+SGN                                                 00004820
 89   GOTO(2000,2001,2311,2321,2331,2341,331,219),N1            00004830
C     TO WRITE SIA 8 FOR MONO STEREO CONTROL                    00004840
 90   P(1)=12.                                                  00004850
      P(3)=8.                                                   00004860
      P(4)=STER                                                 00004870
      IP(1)=4                                                   00004880
      END=0.                                                    00004890
      SNAB=0.                                                   00004900
      GOTO50                                                    00004910
C     FOR PREMATURE END OF FILE ON INPUT                        00004920
 95   NP=2                                                      00004930
      IP(2)=1                                                   00004940
      L=0                                                       00004950
      IBCD(1)=IBC(1)                                            00004960
      GOTO600                                                   00004970
C     TO INITIALIZE                                             00004980
      ENTRYREADD                                                00004990
      CALLFLGEOF(5,IEOF)                                        00005000
      READ1,(CARD(I),I=1,NC)                                    00005010
      PRINT1,(CARD(I),I=1,NC)                                   00005020
      IF(IEOF.EQ.1)GOTO95                                       00005030
      CALLMOVR(CARD,NC)                                         00005040
      P(2)=0.                                                   00005050
      I=0                                                       00005060
      IDEF=0                                                    00005070
      IBK=2                                                     00005080
      STER=0.                                                   00005090
      END=0.                                                    00005100
      SNAB=0.                                                   00005110
      RETURN                                                    00005120
      END                                                       00005130
```

```
*****W  1470 EQUALITY OR NON-EQUALITY COMPARISON MAY NOT BE MEANINGFUL IN LOGICAL IF EXPRESSIONS
*****W     7 MEMORY EXPANDED. USE $LIMITS OR CORE= OPTION FOR NEXT RUN
```

```
      CPASS2              PASS 2 MAIN PROGRAM                                  00005140
      C                     *** MUSIC V ***                                    00005150
            DIMENSIONG(1000),I(1000),T(1000),D(10000),P(100),IP(10)            00005160
            COMMONIP,P,G,I,T,D,IXJQ,TLAST,DLAST                                00005170
      C     INIALIZING PROGRAM                                                 00005180
      C           NOMINAL SAMPLING RATE, NOTE PARAMETER LENGTH, NUMBER OF CARDS 00005190
      C           NO OF OP CODES, PASS II REPORT PRINT PARAMETER               00005200
            G(4)=10000.0                                                       00005210
            NPAR=10000                                                         00005220
            NCAR=1000                                                          00005230
            NOPC=12                                                            00005240
            IXJQ=0                                                             00005250
            IEND=0                                                             00005260
            REWIND10                                                           00005270
            REWIND11                                                           00005280
      C     INIALIZE SECTION                                                   00005290
       150  ID=1                                                               00005300
            IN=1                                                               00005310
            TLAST=0.                                                           00005320
            DLAST=0.                                                           00005330
      C     READ SECTION OF DATA                                               00005340
       106  CALLREAD2(10)                                                      00005350
            I1=IP(1)                                                           00005360
            D(ID)=I1                                                           00005370
            I(IN)=ID                                                           00005380
            T(IN)=P(2)                                                         00005390
            DO100I2=1,I1                                                       00005400
            I3=ID+I2                                                           00005410
       100  D(I3)=P(I2)                                                        00005420
            ID=ID+I1+1                                                         00005430
            IF(ID-NPAR)102,102,101                                             00005440
       101  CALLERROR(20)                                                      00005450
            STOP                                                               00005460
       102  IN=IN+1                                                            00005470
            IF(IN-NCAR)103,103,101                                             00005480
       103  IF(P(1)-5.0)104,110,104                                            00005490
       104  IF(P(1)-6.0)106,105,106                                            00005500
       105  IEND=1                                                             00005510
            GOTO110                                                            00005520
      C     SORT SECTION                                                       00005530
       110  CALLSORTFL                                                         00005540
            IN=IN-1                                                            00005550
            CALLSORT(T(1),T(2),IN,1)                                           00005560
            GOTO120                                                            00005570
      C     EXECUTE OP CODES M SECTION                                         00005580
       120  DO114=1,IN                                                         00005590
            I5=I(I4)                                                           00005600
            I6=D(I5+1)                                                         00005610
            IF(I6)121,121,122                                                  00005620
       121  CALLERROR(21)                                                      00005630
            GOTO1                                                              00005640
       122  IF(I6-NOPC)123,123,121                                             00005650
```

```
      123  GOTO(2,2,2,2,2,2,7,8,7,10,2,6),16                    00005660
      7    CALLERROR(22)                                        00005670
           GOTO1                                                00005680
      8    I7=D(I5)                                             00005690
           I8=I5+4                                              00005700
           I9=I5+I7                                             00005710
           I10=IFIX(D(I5+3))-I8                                 00005720
           DO124I11=I8,I9                                       00005730
           I12=I10+I11                                          00005740
      124  G(I12)=D(I11)                                        00005750
           IF(I6-12)1,2,1                                       00005760
      10   I13=D(I5+3)                                          00005770
           IP(2)=I5                                             00005780
           IF(I13)125,125,126                                   00005790
      125  CALLERROR(23)                                        00005800
           GOTO1                                                00005810
      126  IF(I13-5)127,127,125                                 00005820
      127  GOTO(21,22,23,24,25),I13                             00005830
      21   CALLPLS1                                             00005840
           GOTO1                                                00005850
      22   CALLPLS2                                             00005860
           GOTO1                                                00005870
      23   CALLPLS3                                             00005880
           GOTO1                                                00005890
      24   CALLPLS4                                             00005900
           GOTO1                                                00005910
      25   CALLPLS5                                             00005920
           GOTO1                                                00005930
      C    WRITE OUT SECTION                                    00005940
      2    IP(1)=D(I5)                                          00005950
           I18=IP(1)                                            00005960
           DO133I19=1,I18                                       00005970
           I20=I19+I5                                           00005980
      133  P(I19)=D(I20)                                        00005990
           CALLWRITE2(11)                                       00006000
      1    CONTINUE                                             00006010
      C    END SECTION OR PASS                                  00006020
      140  IF(IEND)141,141,143                                  00006030
      141  PRINT142                                             00006040
      142  FORMAT(22HEND OF SECTION PASS II)                    00006050
           GOTO150                                              00006060
      143  PRINT144                                             00006070
      144  FORMAT(14HEND OF PASS II)                            00006080
           STOP                                                 00006090
           END                                                  00006100
```

```
      SUBROUTINE PLS1                                                   00006110
      COMMON IP,P,G,I,T,D                                               00006120
      DIMENSION IP(10),P(100),G(1000),I(1000),                          00006130
     2T(1000),D(10000)                                                  00006140
      I1=IP(2)                                                          00006150
      IN=IP(3)                                                          00006160
      NQ=D(I1+4)                                                        00006170
      NB=NQ+1                                                           00006180
      NL=NQ+IFIX(G(NQ))                                                 00006190
      DO 103 J=1,IN                                                     00006200
      ID=I(J)                                                           00006210
      IF(D(ID+1)-1.0) 103,100,103                                       00006220
  103 FREQ=D(ID+6)                                                      00006230
      MIN=1000000.0                                                     00006240
      DO 102 K=NB,NL                                                    00006250
      IF(ABS(FREQ-G(K))-MIN) 101,102,102                                00006260
  101 MIN=ABS(FREQ-G(K))                                                00006270
      QFREQ=G(K)                                                        00006280
  102 CONTINUE                                                          00006290
      D(ID+6)=QFREQ                                                     00006300
  103 CONTINUE                                                          00006310
      RETURN                                                            00006320
      END                                                               00006330
```

```
 1          SUBROUTINE CONVT                                          00006340
 2          COMMON IP,P,G                                             00006350
 3          DIMENSION IP(10),P(100),G(1000)                           00006360
 4          IF(G(9).NE.2.0) GO TO 103                                 00006370
 5          IF(P(1)-1.0) 102,101,102                                  00006380
 6     101  P(5)=10.0**(P(5)/20.0)                                    00006390
 7          P(7)=511.0*P(6)/G(4)                                      00006400
 8          P(6)=511.0/(P(4)*G(4))                                    00006410
 9          IP(1)=7                                                   00006420
10     102  RETURN                                                   00006430
11     103  IF(G(9).NE.3.0) GO TO 107                                00006440
12          IF(P(1)-1.0) 104,106,104                                  00006450
13     106  P(5)=10.0**(P(5)/20.0)                                    00006460
14          P(7)=511.0*262.0*(2.0**(P(6)*P(7)/12.0))/G(4)             00006470
15          P(6)=511.0/(P(4)*G(4))                                    00006480
16          P(8)=P(7)*G(50)/100.0                                     00006490
17          P(9)=G(51)*511.0/G(4)                                     00006500
18          IP(2)=9                                                   00006510
19     104  RETURN                                                   00006520
20     107  IF(G(9).NE.4.0) GO TO 111                                00006530
21          IF(P(1)-1.0) 110,109,110                                  00006540
22     109  P(5)=10.0**(P(5)/20.0)                                    00006550
23          P(7)=G(50)*511.0/G(4)                                     00006560
24          P(8)=(P(6)-G(52))*511.0/G(4)                              00006570
25          G(52)=P(6)                                                00006580
26          P(6)=511.0/(P(4)*G(4))                                    00006590
27          P(9)=P(6)                                                 00006600
28          IP(2)=9                                                   00006610
29     110  RETURN                                                   00006620
30     111  IF(G(9).NE.5.0) RETURN                                   00006630
31          IF(P(1)-1.0) 117,113,117                                  00006640
32     113  COR=1.0                                                  00006650
33          SS=P(2)-G(50)-G(51)                                       00006660
34          IF(SS) 114,115,115                                        00006670
35     114  COR=P(4)/G(50)+G(51)                                     00006680
36          P(10)=128.                                                00006690
37          GO TO 116                                                 00006700
38     115  P(10)=128./(G(4)*SS)                                      00006710
39     116  P(9)=128./(G(4)*G(50)*COR)                               00006720
40          P(11)=128./(G(4)*G(51)*COR)                               00006730
41          P(5)=10.0**(P(5)/20.0)                                    00006740
42          P(7)=511.0*P(6)/G(4)                                      00006750
43          P(8)=.0075*P(7)                                           00006760
44          IP(1)=11                                                  00006770
45     117  RETURN                                                   00006780
46          END                                                       00006790
*****W  1470 EQUALITY OR NON-EQUALITY COMPARISON MAY NOT BE MEANINGFUL IN LOGICAL IF EXPRESSIONS
```

```
CCON2          PASS 2 FUNCTION INTERPOLATER                          00006800
C                 *** MUSIC V ***                                    00006810
      FUNCTION CON(G,I,T)                                            00006820
      DIMENSION G(I)                                                 00006830
      DO 10 J=I,1000,2                                               00006840
      IF (G(J)-T) 10,20,30                                           00006850
   30 CON = G(J-1)+((T-G(J-2))/(G(J)-G(J-2)))*(G(J+1)-G(J-1))        00006860
      RETURN                                                         00006870
   10 CONTINUE                                                       00006880
   20 CON = G(J+1)                                                   00006890
      RETURN                                                         00006900
      END                                                            00006910
```

```
      CWRIT2          DATA OUTPUTING ROUTINE FOR PASS 2                 00006920
      C                 *** MUSIC V ***                                 00006930
            SUBROUTINE WRITE2(N)                                        00006940
            COMMON IP(10),P(100),G(1000),I(1000),T(1000),D(10000),IXJQ,TLAST,B  00006950
           1LAST                                                        00006960
            IF(G(2).EQ.0.)GO TO150                                      00006970
            X=P(2)                                                      00006980
            Y=P(4)                                                      00006990
            ILOC=G(2)                                                   00007000
            IF(P(1).NE.1.)GO TO50                                       00007010
            P(4)=P(4)*60./CON(G,ILOC,P(2))                              00007020
      50    P(2)=TLAST+(P(2)-BLAST)*60./CON(G,ILOC,P(2))                00007030
            TLAST=P(2)                                                  00007040
            BLAST=X                                                     00007050
      150   CALL CONVT                                                  00007060
            K=IP(1)                                                     00007070
            WRITE(N)K,(P(J),J=1,K)                                      00007080
      C     *** PASS II REPORT IS OPTIONAL ***                          00007090
            IF(G(1).NE.0.)RETURN                                        00007100
            IF(IXJQ.EQ.0)PRINT100                                       00007110
            IXJQ=10                                                     00007120
      100   FORMAT(15H1PASS II REPORT/11HO(WORD CN1))                   00007130
            PRINT101,K,(P(J),J=1,K)                                     00007140
            IF(G(2).NE.0.)PRINT102,X,Y                                  00007150
      101   FORMAT(I8,10(F9.3))                                         00007160
      102   FORMAT(1H+,110X,2HB=,F7.4,2HD=,F7.4)                        00007170
            RETURN                                                      00007180
            END                                                         00007190
******W  1470 EQUALITY OR NON-EQUALITY COMPARISON MAY NOT BE MEANINGFUL IN LOGICAL IF EXPRESSIONS
```

```
      CSORT                 SORTING PROGRAM                                  00007200
      C                     *** MUSIC V ***                                  00007210
            SUBROUTINE SORT(A,B,N,L)                                         00007220
            DIMENSION A(N),L(N)                                              00007230
      C                                                                      00007240
      C     SORT SORTS THE A ARRAY INTO ASCENDING NUMERICAL ORDER, PERFORMING 00007250
      C     THE SAME OPERATIONS ON ARRAY L AS ON A                           00007260
      C                                                                      00007270
            N1=N-1                                                           00007280
            DO 10 I=1,N1                                                     00007290
            IN=I+1                                                           00007300
            DO 20 J=IN,N                                                     00007310
            IF(A(I).LE.A(J))GO TO 20                                         00007320
            T=A(I)                                                           00007330
            A(I)=A(J)                                                        00007340
            A(J)=T                                                           00007350
            NT=L(I)                                                          00007360
            L(I)=L(J)                                                        00007370
            L(J)=NT                                                          00007380
      20    CONTINUE                                                         00007390
      10    CONTINUE                                                         00007400
            RETURN                                                           00007410
            ENTRY SORTFL                                                     00007420
            RETURN                                                           00007430
            END                                                              00007440
```

```
CREAD2              PASS 2 DATA INPUT ROUTINE                    00007450
C                    *** MUSIC V ***                             00007460
      SUBROUTINEREAD2(N)                                         00007470
      DIMENSIONIP(10),P(100)                                     00007480
      COMMONIP,P                                                 00007490
      READ(N)K,(P(J),J=1,K)                                      00007500
      IP(1)=K                                                    00007510
      RETURN                                                     00007520
      END                                                        00007530
```

```
      CPASS3              PASS 3 MAIN PROGRAM                                00007540
      C                     *** MUSIC V ***                                  00007550
      *       DATA SPECIFICATION                                             00007560
              DIMENSION I(15000),P(100),IP(20),T(50),TI(50),III(53)          00007570
              COMMON I,P/PARM/IP                                             00007580
              DATA IRAN/02736731631551/                                      00007590
      *       INIALIZATION OF PIECE                                          00007600
              I(7)=IRAN                                                      00007610
              IP9=IP(9)                                                      00007620
              REWIND 10                                                      00007630
              REWIND 11                                                      00007640
              SCLFT=IP(12)                                                   00007650
              CALLFROUT0(66,320)                                             00007660
              I(2)=IP(4)                                                     00007670
              MS1=IP(7)                                                      00007680
              MS3=MS1+(IP(8)*IP(9))-1                                        00007690
              MS2=IP(8)                                                      00007700
              I(4)=IP(3)                                                     00007710
              MOUT=IP(10)                                                    00007720
      *       INIALIZATION OF SECTION                                        00007730
       222    T(1)=0.0                                                       00007740
              DO220N1=MS1,MS3,MS2                                            00007750
       220    I(N1)=-1                                                       00007760
              DO221N1=1,IP9                                                  00007770
       221    II(N1)=1000000.                                                00007780
      *       MAIN CARD READING LOOP                                         00007790
       204    CALLDATA                                                       00007800
              IF(P(2)-T(1))200,200,244                                       00007810
       200    IOP=P(1)                                                       00007820
              IF(IOP)201,201,202                                             00007830
       201    CALLERROR(1)                                                   00007840
              GOTO204                                                        00007850
       202    IF(IP(1)-IOP)201,203,203                                       00007860
       203    GOTO(1,2,3,4,5,6,201,201,201,201,11,11),IOP                    00007870
       11     IVAR=P(3)                                                      00007880
              IVARE=IVAR+I(1)-4                                              00007890
              DO 297 N1=IVAR,IVARE                                           00007900
              IVARP=N1-IVAR+4                                                00007910
       297    I(N1)=P(IVARP)                                                 00007920
              GOTO204                                                        00007930
       3      IGEN=P(3)                                                      00007940
              GOTO(281,282,283,284,285),IGEN                                 00007950
       281    CALLGEN1                                                       00007960
              GOTO204                                                        00007970
       282    CALLGEN2                                                       00007980
              GOTO204                                                        00007990
       283    CALLGEN3                                                       00008000
              GOTO204                                                        00008010
       284    CALLGEN4                                                       00008020
              GOTO204                                                        00008030
       285    CALLGEN5                                                       00008040
              GOTO204                                                        00008050
```

```
      4     IVAR=P(3)                                                00008060
            IVARE=IVAR+I(1)-4                                        00008070
            DO296N1=IVAR,IVARE                                       00008080
            IVARP=N1-IVAR+4                                          00008090
      296   I(N1+100)=P(IVARP)+SCLFT                                 00008100
            GOTO204                                                  00008110
      5     GOTO222                                                  00008120
      6     CALLFROUT3                                               00008130
            STOP                                                     00008140
      *     ENTER NOTE TO BE PLAYED                                  00008150
      1     DO230M1=MS1,MS3,MS2                                      00008160
            IF(I(NT)+1)230,231,230                                   00008170
      230   CONTINUE                                                 00008180
            CALLERROR(2)                                             00008190
            GOTO204                                                  00008200
      231   M1=N1                                                    00008210
            M2=N1+I(1)-1                                             00008220
            M3=M2+1                                                  00008230
            M4=N1+IP(8)-1                                            00008240
            DO232M1=M1,M2                                            00008250
            M5=N1-M1+1                                               00008260
      232   I(N1)=P(M5)+SCLFT                                        00008270
            I(M1  )=P(3)                                             00008280
            DO233N1=M3,M4                                            00008290
      233   I(N1)=0                                                  00008300
            DO235N1=1,IP9                                            00008310
            IF(TI(N1)-1000000.)235,234,235                           00008320
      234   II(N1)=P(2)+P(4)                                         00008330
            ITI(N1)=M1                                               00008340
            GOTO204                                                  00008350
      235   CONTINUE                                                 00008360
            CALLERROR(3)                                             00008370
            GOTO204                                                  00008380
      *     DEFINE INSTRUMENT                                        00008390
      2     M1=I(2)                                                  00008400
            M2=IP(5)+IFIX(P(3))                                      00008410
            I(M2)=M1                                                 00008420
      218   CALLDATA                                                 00008430
            IF(I(1)-2)210,210,211                                    00008440
      210   I(M1)=0                                                  00008450
            I(2)=M1+1                                                00008460
            GOTO204                                                  00008470
      211   I(M1)=P(3)                                               00008480
            M3=I(1)                                                  00008490
            I(M1+1)=M1+M3-1                                          00008500
            M1=M1+2                                                  00008510
            DO217N1=4,M3                                             00008520
            M5=P(N1)                                                 00008530
            IF(M5)212,213,213                                        00008540
      212   IF(M5+100)300,301,301                                    00008550
      300   I(M1)=-IP(2)+(M5+101)*IP(6)                              00008560
            GOTO216                                                  00008570
```

```
      331   I(M1)=-IP(13)+(M5+1)+IP(14)                                00008580
            GOTO216                                                    00008590
      213   IF(M5-IP(8))214,214,215                                    00008600
      214   I(M1)=M5                                                   00008610
            GOTO216                                                    00008620
      215   I(M1)=M5+262144                                            00008630
      216   M1=M1+1                                                    00008640
      217   CONTINUE                                                   00008650
            GOTO218                                                    00008660
      *     PLAY TO ACTION TIME                                        00008670
      244   T(2)=P(2)                                                  00008680
      250   TMIN=1000000.                                              00008690
            IREST=1                                                    00008700
            DO241N1=1,IP9                                              00008710
            IF(TMIN-TI(N1))241,241,240                                 00008720
      240   TMIN=TI(N1)                                                00008730
            MNOTE=N1                                                   00008740
      241   CONTINUE                                                   00008750
            IF(1000000.-TMIN)251,251,243                               00008760
      243   IF(TMIN-T(2))245,245,246                                   00008770
      245   T(3)=TMIN                                                  00008780
            GOTO260                                                    00008790
      246   T(3)=T(2)                                                  00008800
            GOTO260                                                    00008810
      247   IF(T(1)-T(2))249,200,200                                   00008820
      249   TI(MNOTE)=1000000.                                         00008830
            M2=ITI(MNOTE)                                              00008840
            I(M2)=-1                                                   00008850
            GOTO250                                                    00008860
      *     SETUP REST                                                 00008870
      251   T(3)=T(2)                                                  00008880
            IREST=2                                                    00008890
            GOTO260                                                    00008900
      *     PLAY                                                       00008910
      260   ISAM=(T(3)-T(1))*FLOAT(I(4))+.5                            00008920
            I(1)=I(3)                                                  00008930
            IF(ISAM)247,247,266                                        00008940
      266   IF(ISAM-IP(14))262,262,263                                 00008950
      262   I(5)=ISAM                                                  00008960
            ISAM=0                                                     00008970
            GOTO264                                                    00008980
      263   I(5)=IP(14)                                                00008990
            ISAM=ISAM-IP(14)                                           00009000
      264   IF(I(8))290,290,291                                        00009010
      290   M3=MOUT+I(5)-1                                             00009020
            MSAMP=I(5)                                                 00009030
            GOTO292                                                    00009040
      291   M3=MOUT+(2*I(5))-1                                         00009050
            MSAMP=2*I(5)                                               00009060
      292   DO267N1=MOUT,M3                                            00009070
      267   I(N1)=0                                                    00009080
            GOTO(268,265),IREST                                        00009090
```

```
      268  DO270MS1=MS1,MS3,MS2                                  00009100
           IF(I(MS1)+1)269,270,269                               00009110
      269  GOTO271                                               00009120
      270  CONTINUE                                              00009130
      265  CALL SAMOUT(I(MOUT),MSAMP)                            00009140
           IF(ISAM)247,247,266                                   00009150
      *    GO THROUGH UNIT GENERATORS IN INSTRUMENT              00009160
      271  I(3)=MS1                                              00009170
           IGEN=IP(5)+I(MS1)                                     00009180
           IGEN=I(IGEN)                                          00009190
      272  I(6)=IGEN                                             00009200
           IF(I(IGEN)-101)293,294,294                            00009210
      293  CALLSAMGEN(I)                                         00009220
           GOTO295                                               00009230
      294  CALLFORSAM                                            00009240
      295  IGEN=I(IGEN+1)                                        00009250
           IF(I(IGEN))270,270,272                                00009260
           END                                                   00009270
```

```
      CGEN1                FUNCTION GENERATOR 1                          00009280
      C                     *** MUSIC V ***                              00009290
            SUBROUTINEGEN1                                               00009300
            DIMENSION I(15000),P(100),IP(20)                             00009310
            COMMON I,P/PARM/IP                                           00009320
            N1=IP(2)+(IFIX(P(4))-1)*IP(6)                                00009330
            M1=7                                                         00009340
            SCLFT=IP(15)                                                 00009350
      102   IF(P(M1+1))103,103,100                                       00009360
      100   V1=P(M1-2)*SCLFT                                             00009370
            V2=(P(M1)-P(M1-2))/(P(M1+1)-P(M1-1))*SCLFT                   00009380
            MA=N1+IFIX(P(M1-1))                                          00009390
            MB=N1+IFIX(P(M1+1))-1                                        00009400
            DO101J=MA,MB                                                 00009410
            XJ=J-MA                                                      00009420
      101   I(J)=V1+V2*XJ                                                00009430
            IF(IFIX(P(M1+1)).EQ.(IP(6)-1))GOTO103                        00009440
            M1=M1+2                                                      00009450
            GOTO102                                                      00009460
      103   I(MB+1)=P(M1)*SCLFT                                          00009470
            RETURN                                                       00009480
            END                                                          00009490
```

```
CGEN2                FUNCTION GENERATOR 2                                00009500
C                      *** MUSIC V ***                                   00009510
      SUBROUTINEGEN2                                                     00009520
      DIMENSION I(15000),P(100),IP(20),A(7000)                           00009530
      COMMONI,P/PARM/IP                                                  00009540
      EQUIVALENCE(I,A)                                                   00009550
      SCLFT=IP(15)                                                       00009560
      N1=IP(2)+(IFIX(P(4))-1)*IP(6)                                      00009570
      N2=N1+IP(6)-1                                                      00009580
      DO101K1=N1,N2                                                      00009590
  101 A(K1)=0.0                                                          00009600
      FAC=6.283185/(FLOAT(IP(6))-1.0)                                    00009610
      NMAX=I(1)                                                          00009620
      N3=5+INT(ABS(P(NMAX)))-1                                           00009630
      IF(N3-5)104,100,100                                                00009640
  100 DO103J=5,N3                                                        00009650
      FACK=FAC*FLOAT(J-4)                                                00009660
      DO102K=N1,N2                                                       00009670
  102 A(K)=A(K)+SIN(FACK*FLOAT(K-N1))*P(J)                               00009680
  103 CONTINUE                                                           00009690
  104 N4=N3+1                                                            00009700
      N5=I(1)-1                                                          00009710
      IF(N5-N4)108,105,105                                               00009720
  105 DO107J1=N4,N5                                                      00009730
      FACK=FAC*FLOAT(J1-N4)                                              00009740
      DO106K1=N1,N2                                                      00009750
  106 A(K1)=A(K1)+COS(FACK*FLOAT(K1-N1))*P(J1)                           00009760
  107 CONTINUE                                                           00009770
      IF(P(NMAX))112,112,108                                             00009780
  108 FMAX=0.0                                                           00009790
      DO110K2=N1,N2                                                      00009800
      IF(ABS(A(K2))-FMAX)110,110,109                                     00009810
  109 FMAX=ABS(A(K2))                                                    00009820
  110 CONTINUE                                                           00009830
  113 DO111K3=N1,N2                                                      00009840
  111 I(K3)=(A(K3)*SCLFT+.99999)/FMAX                                    00009850
      RETURN                                                             00009860
  112 FMAX=.99999                                                        00009870
   GO TO 113
      END                                                                00009890
```

```
      CGEN3              FUNCTION GENERATOR 3                                  00009900
      C                   *** MUSIC V ***                                      00009910
      C      ASSUMPTIONS--P(4) = THE NUMBER OF THE FUNCTION TO BE GENERATED,   00009920
      C      I(1) = WORD COUNT FOR CURRENT DATA RECORD                         00009930
      C      P(5)  = THE BEGINNING THE THE LIST OF DESCRIPTION NUMBERS         00009940
      C      IP(2) = THE BEGINNING SUBSCRIPT FOR FUNCTIONS IN THE I ARRAY,     00009950
      C      IP(6) = THE LENGTH OF THE FUNCTIONS                               00009960
      C      IP(15) = SCALE FACTOR FOR STORED FUNCTIONS                        00009970
      C                                                                        00009980
             SUBROUTINE GEN3                                                   00009990
             COMMON I(15000),P(100) /PARM/ IP(20)                              00010000
             N=I(1)-5                                                          00010010
             NL=5                                                              00010020
             SCLFT=IP(15)                                                      00010030
             LL=IP(6)                                                          00010040
             RMIN=0                                                            00010050
             RMAX=0                                                            00010060
             NR=NL+N                                                           00010070
             DO 10 J=NL,NR                                                     00010080
             IF(P(J).GT.RMAX) RMAX=P(J)                                        00010090
       10    IF(P(J).LT.RMIN) RMIN=P(J)                                        00010100
             DIV=AMAX1(ABS(RMIN),ABS(RMAX))                                    00010110
             N1 = IP(2) + (IFIX(P(4))-1)*IP(6)                                 00010120
             I(N1)=(P(NL)/DIV)*SCLFT                                           00010130
             LAST = N1                                                         00010140
             DO 100 J=1,N                                                      00010150
             LL = LL-LL/(N-J+1)                                                00010160
             IX = N1+IP(6)-LL-1                                                00010170
             IX2 = NL+J                                                        00010180
             I(IX)=(P(IX2)/DIV)*SCLFT                                          00010190
             DELTA=FLOAT(I(IX))-FLOAT(I(LAST))                                 00010200
             NR = IX-LAST-1                                                    00010210
             SEG = NR+1                                                        00010220
             HNCR=DELTA/SEG                                                    00010230
             DO 50 K=1,NR                                                      00010240
             IX2 = LAST+K                                                      00010250
       50    I(IX2)=FLOAT(I(IX2-1))+HNCR                                       00010260
      100    LAST=IX                                                           00010270
             RETURN                                                            00010280
             END                                                               00010290
```

```
1     CDATA3          PASS 3 DATA INPUTING ROUTINE                    00010300
2     C                 *** MUSIC V ***                               00010310
3           SUBROUTINE DATA                                           00010320
4           COMMON I(15000),P(100)                                    00010330
5           READ (11) K,(P(J),J=1,K)                                  00010340
6           I(1)=K                                                    00010350
7           RETURN                                                    00010360
8           END                                                       00010370
```

```
CFORS3              FORTRAN UNIT GENERATOR ROUTINE                          00010380
C                     *** MUSIC V ***                                       00010390
      SUBROUTINEFORSAM                                                      00010400
      DIMENSION I(15000),P(100),IP(20),L(8),M(8)                            00010410
      COMMONI,P/PARM/IP                                                     00010420
      EQUIVALENCE(M1,M(1)),(M2,M(2)),(M3,M(3)),(M4,M(4)),(M5,M(5)),(M6,M    00010430
     1(6)),(M7,M(7)),(M8,M(8)),(L1,L(1)),(L2,L(2)),(L3,L(3)),(L4,L(4)),(    00010440
     2L5,L(5)),(L6,L(6)),(L7,L(7)),(L8,L(8)),(RN1,IRN1),(RN3,IRN3),(RN,I    00010450
     3RN)                                                                   00010460
C     MULTIPLIER FOR RANDOM NUMBERS:CF. J.A.C.M. 1967 PP100-119             00010470
      DATA IMULT/0273673163155/                                             00010480
      DATA MASK/0377777777777/                                              00010490
      SFI=1./FLOAT(IP(12))                                                  00010500
      SFF=1./FLOAT(IP(15))                                                  00010510
      SFID=FLOAT(IP(12))                                                    00010520
      SFXX=FLOAT(IP(12))/FLOAT(IP(15))                                      00010530
      XNFUN=IP(6)-1                                                         00010540
*     COMMON INITIALIZATION OF GENERATORS                                   00010550
      N1=I(6)+2                                                             00010560
      N2=I(N1-1)-1                                                          00010570
      DO204J1=N1,N2                                                         00010580
      J2=J1-N1+1                                                            00010590
      IF(I(J1))200,201,201                                                  00010600
  200 L(J2)=-I(J1)                                                          00010610
      M(J2)=1                                                               00010620
      GOTO204                                                               00010630
  201 M(J2)=0                                                               00010640
      IF(I(J1)-262144)202,202,203                                           00010650
  202 L(J2)=I(J1)+I(3)-1                                                    00010660
      GOTO204                                                               00010670
  203 L(J2)=I(J1)-262144                                                    00010680
  204 CONTINUE                                                              00010690
      NSAM=I(5)                                                             00010700
      N3=I(N1-2)                                                            00010710
      NGEN=  N3 -100                                                        00010720
      GOTO(101,102,103,104,105,106,107,108,109,110,111,112,                 00010730
     2112,112,970),NGEN                                                     00010740
  112 RETURN                                                                00010750
*     UNIT GENERATORS                                                       00010760
C     OUTPUT BOX                                                            00010770
  101 IF(M1)260,260,261                                                     00010780
  260 IN1=I(L1)                                                             00010790
  261 CONTINUE                                                              00010800
      DO270J3=1,NSAM                                                        00010810
      IF(M1)265,265,264                                                     00010820
  264 J4=L1+J3-1                                                            00010830
      IN1=I(J4)                                                             00010840
  265 J5=L2+J3-1                                                            00010850
      I(J5)=IN1+I(J5)                                                       00010860
  270 CONTINUE                                                              00010870
      RETURN                                                                00010880
C     OSCILLATOR                                                            00010890
```

```
      102   SUM=FLOAT(I(L5))*SFI                                    00010900
            IF(M1)280,280,281                                       00010910
      280   AMP=FLOAT(I(L1))*SFI                                    00010920
      281   IF(M2)282,282,283                                       00010930
      282   FREQ=FLOAT(I(L2))*SFI                                   00010940
      283   CONTINUE                                                00010950
            DO293J3=1,NSAM                                          00010960
            J4=INT(SUM)+L4                                          00010970
            F=FLOAT(I(J4))                                          00010980
            IF(M2)285,285,286                                       00010990
      285   SUM=SUM+FREQ                                            00011000
            GOTO290                                                 00011010
      286   J4=L2+J3-1                                              00011020
            SUM=SUM+FLOAT(I(J4))*SFI                                00011030
      290   IF(SUM-XMFUN)288,287,287                                00011040
      287   SUM=SUM-XMFUN                                           00011050
      288   J5=L3+J3-1                                              00011060
            IF(M1)291,291,292                                       00011070
      291   I(J5)=IFIX(AMP*F*SFXX)                                  00011080
            GOTO293                                                 00011090
      292   J6=L1+J5-1                                              00011100
            I(J5)=IFIX(FLOAT(I(J6))*F*SFF)                          00011110
      293   CONTINUE                                                00011120
            I(L5)=IFIX(SUM*SFID)                                    00011130
            RETURN                                                  00011140
C           ADD TWO BOX                                             00011150
      103   IF(M1)250,250,251                                       00011160
      250   IN1=I(L1)                                               00011170
      251   IF(M2)252,252,253                                       00011180
      252   IN2=I(L2)                                               00011190
      253   DO258J3=1,NSAM                                          00011200
            IF(M1)255,255,254                                       00011210
      254   J4=L1+J3-1                                              00011220
            IN1=I(J4)                                               00011230
      255   IF(M2) 257,257,256                                      00011240
      256   J5=L2+J3-1                                              00011250
            IN2=I(J5)                                               00011260
      257   J6=L3+J3-1                                              00011270
            I(J6)=IN1+IN2                                           00011280
      258   CONTINUE                                                00011290
            RETURN                                                  00011300
C           RANDOM INTERPOLATING GENERATOR                          00011310
      104   SUM=FLOAT(I(L4))*SFI                                    00011320
            IF(M1)310,310,311                                       00011330
      310   XIN1=FLOAT(I(L1))*SFI                                   00011340
      311   IF(M2)312,312,313                                       00011350
      312   XIN2=FLOAT(I(L2))*SFI                                   00011360
      313   IRN1=I(L5)                                              00011370
            IRN3=I(L6)                                              00011380
            DO340J3=1,NSAM                                          00011390
            IF(M1)316,316,315                                       00011400
      315   J4=L1+J3-1                                              00011410
```

```
      XIN1=FLOAT(I(J4))*SFI                                     00011420
  316 IF(M2)318,318,317                                         00011430
  317 J5=L2+J3-1                                                00011440
      XIN2=FLOAT(I(J5))*SFI                                     00011450
  318 IF(SUM-XNFUN)320,319,319                                  00011460
  319 SUM=SUM-XNFUN                                             00011470
      I(7)=I(7)*IMULT                                           00011480
      I(7)=AND(I(7),MASK)                                       00011490
      RN4=(2.*FLOAT(I(7))*SFF-1.)                               00011500
      RN2=RN4-RN3                                               00011510
      RN1=RN3                                                   00011520
      RN3=RN4                                                   00011530
      GOTO321                                                   00011540
  320 RN2=RN3-RN1                                               00011550
  321 J7=L3+J3-1                                                00011560
      I(J7)=XIN1*(RN1+(RN2*SUM)/XNFUN)*SFID                     00011570
      SUM=SUM+XIN2                                              00011580
  340 CONTINUE                                                  00011590
      I(L4)=IFIX(SUM*SFID)                                      00011600
      I(L5)=IRN1                                                00011610
      I(L6)=IRN3                                                00011620
      RETURN                                                    00011630
C     ENVELOPE GENERATOR                                        00011640
  105 SUM=FLOAT(I(L7))*SFI                                      00011650
      IF(M1)380,380,381                                         00011660
  380 XIN1=FLOAT(I(L1))*SFI                                     00011670
  381 IF(M4)382,382,383                                         00011680
  382 XIN4=FLOAT(I(L4))*SFI                                     00011690
  383 IF(M5)384,384,385                                         00011700
  384 XIN5=FLOAT(I(L5))*SFI                                     00011710
  385 IF(M6)386,386,387                                         00011720
  386 XIN6=FLOAT(I(L6))*SFI                                     00011730
  387 X1=XNFUN/4.                                               00011740
      X2=2.*X1                                                  00011750
      X3=3.*X1                                                  00011760
      DO403J3=1,NSAM                                            00011770
      J4=INT(SUM)+L2                                            00011780
      F=FLOAT(I(J4))                                            00011790
      IF(M1)405,405,404                                         00011800
  404 J8=L1+J3-1                                                00011810
      XIN1=FLOAT(I(J8))*SFI                                     00011820
  405 IF(SUM-XNFUN)389,388,388                                  00011830
  388 SUM=SUM-XNFUN                                             00011840
  389 IF(SUM-X1)390,390,393                                     00011850
  390 IF(M4)392,392,391                                         00011860
  391 J4=L4+J3-1                                                00011870
      XIN4=FLOAT(I(J4))*SFI                                     00011880
  392 SUM=SUM+XIN4                                              00011890
      GOTO402                                                   00011900
  393 IF(SUM-X2)394,394,397                                     00011910
  394 IF(M5)396,396,395                                         00011920
  395 J5=L5+J3-1                                                00011930
```

```
      XIN5=FLOAT(I(J5))*SFI                                     00011940
  396 SUM=SUM+XIN5                                              00011950
      GOTO402                                                   00011960
  397 IF(M6)400,400,399                                         00011970
  399 J6=L6+J3-1                                                00011980
      XIN6=FLOAT(I(J6))*SFI                                     00011990
  400 SUM=SUM+XIN6                                              00012000
  402 J7=L3+J3-1                                                00012010
      I(J7)=IFIX(XIN1*F*SFXX)                                   00012020
  403 CONTINUE                                                  00012030
      I(L7)=IFIX(SUM*SFID)                                      00012040
      RETURN                                                    00012050
C     STEREO OUTPUT BOX                                         00012060
  136 IF(M1)500,500,501                                         00012070
  500 IN1=I(L1)                                                 00012080
  501 IF(M2)502,502,503                                         00012090
  502 IN2=I(L2)                                                 00012100
  503 NSSAM=2*NSAM                                              00012110
      DO510J3=1,NSSAM,2                                         00012120
      IF(M1)505,505,504                                         00012130
  504 J4=L1+J3-1                                                00012140
      IN1=I(J4)                                                 00012150
  505 J5=L3+J3-1                                                00012160
      I(J5)=IN1+I(J5)                                           00012170
      IF(M2)507,507,506                                         00012180
  506 J4=L2+J3-1                                                00012190
      IN2=I(J4)                                                 00012200
  507 J5=L3+J3                                                  00012210
      I(J5)=IN2+I(J5)                                           00012220
  510 CONTINUE                                                  00012230
      RETURN                                                    00012240
C     ADD 3 BOX                                                 00012250
  137 IF(M1)750,750,751                                         00012260
  750 IN1=I(L1)                                                 00012270
  751 IF(M2)752,752,753                                         00012280
  752 IN2=I(L2)                                                 00012290
  753 IF(M3)754,754,755                                         00012300
  754 IN3=I(L3)                                                 00012310
  755 DO780J3=1,NSAM                                            00012320
      IF(M1)757,757,756                                         00012330
  756 J4=4+J3-1                                                 00012340
      IN1=I(J4)                                                 00012350
  757 IF(M2)759,759,758                                         00012360
  758 J5=L2+J3-1                                                00012370
      IN2=I(J5)                                                 00012380
  759 IF(M3)761,761,760                                         00012390
  760 J6=L3+J3-1                                                00012400
      IN3=I(J6)                                                 00012410
  761 J7=L4+J3-1                                                00012420
      I(J7)=IN1+IN2+IN3                                         00012430
  780 CONTINUE                                                  00012440
      RETURN                                                    00012450
```

```
C       ADD & BOX                                                       00012460
 108    IF(M1)850,850,851                                               00012470
 850    IN1=I(L1)                                                       00012480
 851    IF(M2)852,852,853                                               00012490
 852    IN2=I(L2)                                                       00012500
 853    IF(M3)854,854,855                                               00012510
 854    IN3=I(L3)                                                       00012520
 855    IF(M4)856,856,857                                               00012530
 856    IN4=I(L4)                                                       00012540
 857    DO880J3=1,MSAM                                                  00012550
        IF(M1)859,859,858                                               00012560
 858    J4=L1+J3-1                                                      00012570
        IN1=I(J4)                                                       00012580
 859    IF(M2)861,861,860                                               00012590
 860    J5=L2+J3-1                                                      00012600
        IN2=I(J5)                                                       00012610
 861    IF(M3)863,863,862                                               00012620
 862    J6=L3+J3-1                                                      00012630
        IN3=I(J6)                                                       00012640
 863    IF(M4)865,865,864                                               00012650
 864    J7=L4+J3-1                                                      00012660
        IN4=I(J7)                                                       00012670
 865    J8=L5+J3-1                                                      00012680
        I(J8)=IN1+IN2+IN3+IN4                                           00012690
 880    CONTINUE                                                        00012700
        RETURN                                                          00012710
C       MULTIPLIER                                                      00012720
 109    IF(M1)900,900,901                                               00012730
 900    XIN1=FLOAT(I(L1))*SFI                                           00012740
 901    IF(M2)902,902,903                                               00012750
 902    XIN2=FLOAT(I(L2))*SFI                                           00012760
 903    DO908J3=1,MSAM                                                  00012770
        IF(M1)905,905,904                                               00012780
 904    J4=L1+J3-1                                                      00012790
        XIN1=FLOAT(I(J4))*SFI                                           00012800
 905    IF(M2)907,907,906                                               00012810
 906    J5=L2+J3-1                                                      00012820
        XIN2=FLOAT(I(J5))*SFI                                           00012830
 907    J6=L3+J3-1                                                      00012840
        I(J6)=XIN1*XIN2*SFID                                            00012850
 908    CONTINUE                                                        00012860
        RETURN                                                          00012870
C       SET NEW FUNCTION IN OSC OR ENV                                  00012880
 110    ILOC=N1+6                                                       00012890
        IF(I(N1+1).EQ.5) ILOC=N1+4                                      00012900
        IN1=I(3)+I(N1)-1                                                00012910
        IIN1=I(IN1)/IP(12)                                              00012920
        IF(IIN1)960,960,955                                             00012930
 955    I(ILOC)=-IP(2)-(IIN1-1)*IP(6)                                   00012940
 960    RETURN                                                          00012950
C       RANDOM AND HOLD GENERATOR                                       00012960
 111    SUM=FLOAT(I(L4))*SFI                                            00012970
```

```
      IF(M1)910,910,911                                                 00012980
910   XIN1=FLOAT(I(L1))*SFI                                             00012990
911   IF(M2)912,912,913                                                 00013000
912   XIN2=FLOAT(I(L2))*SFI                                             00013010
913   IRN=I(L5)                                                         00013020
      DO940J3=1,NSAM                                                    00013030
      IF(M1)916,916,915                                                 00013040
915   J4=L1+J3-1                                                        00013050
      XIN1=FLOAT(I(J4))*SFI                                             00013060
916   IF(M2)918,918,917                                                 00013070
917   J5=L2+J3-1                                                        00013080
      XIN2=FLOAT(I(J5))*SFI                                             00013090
918   IF(SUM-XNFUN)920,919,919                                          00013100
919   SUM=SUM-XNFUN                                                     00013110
      I(7)=I(7)*IMULT                                                   00013120
      I(7)=AND(I(7),MASK)                                               00013130
      RN=(2.*FLOAT(I(7))*SFF-1.)                                        00013140
920   J7=L3+J3-1                                                        00013150
      I(J7)=XIN1*RN*SFID                                                00013160
      SUM=SUM+XIN2                                                      00013170
940   CONTINUE                                                          00013180
      I(L4)=IFIX(SUM*SFID)                                              00013190
      I(L5)=IRN                                                         00013200
      RETURN                                                            00013210
C     LSG                                                               00013220
970   DO 982 J3=1,NSAM                                                  00013230
      IF(M2) 972,972,984                                                00013240
984   J5=L2+J3-1                                                        00013250
      IN2=I(J5)                                                         00013260
      GO TO 974                                                         00013270
972   IN2=I(L2)                                                         00013280
974   IF(M1) 978,978,976                                                00013290
976   J4=L1+J3-1                                                        00013300
      I(J4)=I(J4)+IN2                                                   00013310
      IN1=I(J4)                                                         00013320
      GO TO 980                                                         00013330
978   I(L1)=I(L1)+IN2                                                   00013340
      IN1=I(L1)                                                         00013350
980   J6=L3+J3-1                                                        00013360
      I(J6)=IN1                                                         00013370
982   CONTINUE                                                          00013380
      RETURN                                                            00013390
      END                                                               00013400
```

```
      *                                                                 00013410
      *     SAMOUT--PACKS AND OUTPUTS SAMPLES TO MAGNETIC TAPE          00013420
      *                                                                 00013430
            SUBROUTINE SAMOUT(NS,N)                                     00013440
            DIMENSION I(15000),P(100),IP(20)                            00013450
            INTEGER I(1341),FRMPTR,TOPTR,BYTPTR,SAMPLE,UNDER,           00013460
           2OVER,NBLK,SAMCNT,SECS,CON/04000400004000/                   00013470
            COMMON I,P/PARM/IP                                          00013480
            NSAMP=N                                                     00013490
            FRMPTR=IP(10)                                               00013500
      1050  IF(NSAMP.LT.1) RETURN                                       00013510
            SAMPLE=IFIX(FLOAT(I(FRMPTR))/FLOAT(IP(12)))+2048            00013520
            SAMCNT=SAMCNT+1                                             00013530
            IF(SAMPLE.LE.4095) GO TO 1110                               00013540
            SAMPLE=4095                                                 00013550
            OVER=OVER+1                                                 00013560
            IF(SAMPLE.GE.0) GO TO 1110                                  00013570
            SAMPLE=0                                                    00013580
            UNDER=UNDER+1                                               00013590
      1110  GO TO (1000,1010,1020), BYTPTR                              00013600
      1000  FLD(0,12,T(TOPTR))=FLD(24,12,SAMPLE)                        00013610
            GO TO 1030                                                  00013620
      1010  FLD(12,12,T(TOPTR))=FLD(24,12,SAMPLE)                       00013630
            GO TO 1030                                                  00013640
      1020  FLD(24,12,T(TOPTR))=FLD(24,12,SAMPLE)                       00013650
      1030  BYTPTR=BYTPTR+1                                             00013660
            IF(BYTPTR.LE.3) GO TO 1040                                  00013670
            BYTPTR=1                                                    00013680
            TOPTR=TOPTR+1                                               00013690
            IF(TOPTR.LE.1341) GO TO 1040                                00013700
            WRITE(26) (T(J),J=1,1341)                                   00013710
            NBLK=NBLK+1                                                 00013720
            TOPTR=1                                                     00013730
            BYTPTR=1                                                    00013740
      1040  NSAMP=NSAMP-1                                               00013750
            FRMPTR=FRMPTR+1                                             00013760
            GO TO 1050                                                  00013770
            ENTRY FROUTO(IFC,IWORDS)                                    00013780
            REWIND 26                                                   00013790
            TOPTR=1                                                     00013800
            BYTPTR=1                                                    00013810
            OVER=0                                                      00013820
            UNDER=0                                                     00013830
            NBLK=0                                                      00013840
            SAMCNT=0                                                    00013850
            RETURN                                                      00013860
            ENTRY FROUT3                                                00013870
      1090  GO TO (1060,1070,1080), BYTPTR                              00013880
      1070  FLD(12,12,T(TOPTR))=FLD(24,12,CON)                         00013890
            BYTPTR=BYTPTR+1                                             00013900
            GO TO 1090                                                  00013910
      1080  FLD(24,12,T(TOPTR))=FLD(24,12,CON)                         00013920
```

```
      IOPTR=IOPTR+1                                                     00013930
 1060 IF(IOPTR.GT.1341) GO TO 1100                                      00013940
      I(IOPTR)=CON                                                      00013950
      IOPTR=IOPTR+1                                                     00013960
      GO TO 1060                                                        00013970
 1100 WRITE(26) (I(J),J=1,1341)                                         00013980
      NBLK=NBLK+1                                                       00013990
      REWIND 26                                                         00014000
      WRITE(6,1120) UNDER                                               00014010
 1120 FORMAT(12H THERE WERE ,I11,21H SAMPLES LESS THAN 0.)              00014020
      WRITE(6,1130) OVER                                                00014030
 1130 FORMAT(12H THERE WERE ,I11,19H SAMPLES OVER 4095.)                00014040
      WRITE(6,1140) NBLK                                                00014050
 1140 FORMAT(1H ,I11,38H BLOCKS WERE WRITTEN TO MAGNETIC TAPE.)         00014060
      SECS=SAMCNT/IP(3)                                                 00014070
      WRITE(6,1150) SECS                                                00014080
 1150 FORMAT(15H PLAYING TIME; ,I11,8H SECONDS)                         00014090
      RETURN                                                            00014100
      END                                                               00014110
```

APPENDIX B

Sample Output Report

```
$$     X2669 ENTERED UKIR66 AT 14.478 FROM SYSTEM-0 CD RDR  0-26-01

0001   $      SNUMB    X2669
0002   $$     IDENT    ########$##########*D,R2OGER                                  09/11/77
0003   $$     SELECT   5919KEN/MUSIC1                                                00000020
0004*  $      OPTION   FORTRAN,NOMAP
0005*  $      OBJECT            PASS 1 MAIN PROGRAM                 Y10.458010873PASS1000
0006*  $      DKEND    CONTINUE                                                     PASS1013
0007*  $      OBJECT                 INTERPRETATIVE READING ROUT Y21.450031275READ1000
0008*  $      DKEND                                                                 READ1056
0009*  $      OBJECT            PASS 1 DATA-WRITING ROUTINE       Y21.420031275WRIT1000
0010*  $      DKEND                                                                 WRIT1004
0011*  $      OBJECT            GENERAL ERROR ROUTINE             Y23.750022575ERRO1000
0012*  $      DKEND                                                                 ERRO1004
0013*  $      OBJECT                                              G02.336022875MOVRL000
0014*  $      DKEND                                                                 MOVRL004
0015*  $      OBJECT                                              Y21.445031275PLF10000
0016*  $      DKEND                                                                 PLF10006
0017*  $      OBJECT                                              Y21.400031275PLF20000
0018*  $      DKEND                                                                 PLF20008
0019*  $      OBJECT                                              Y21.340031275PLF30000
0020*  $      DKEND                                                                 PLF30007
0021*  $      OBJECT                                              Y21.409031275PLF40000
0022*  $      DKEND                                                                 PLF40010
0023*  $      OBJECT            PASS 2 FUNCTION INTERPOLATER      Y21.332031275CON20000
0024*  $      DKEND                                                                 CON20005
0025*+$       EXECUTE
0026*  $      FILE     10,X10S,15L
0027*  $      LIMITS   15,13K
0028   $$     SELECT   5919KEN/MUSIC2                                                00002760
0029*  $      OPTION   FORTRAN,NOMAP
0030*  $      OBJECT            PASS 2 MAIN PROGRAM               Y21.425031275PASS2000
0031*  $      DKEND                                                                 PASS2020
0032*  $      OBJECT                                              Y21.380031275CONVT000
0033*  $      DKEND                                                                 CONVT012
0034*  $      OBJECT            PASS 2 FUNCTION INTERPOLATER      Y21.332031275CON20000
0035*  $      DKEND                                                                 CON20005
0036*  $      OBJECT            DATA OUTPUTING ROUTINE FOR PASS   Y21.330031275WRIT2000
0037*  $      DKEND                                                                 WRIT2011
0038*  $      OBJECT              SORTING PROGRAM                 Y21.250031275SORT0000
0039*  $      DKEND                                                                 SORT0006
0040*  $      OBJECT            PASS 2 DATA INPUT ROUTINE         Y21.237031275READ2000
0041*  $      DKEND                                                                 READ2004
0042*  $      OBJECT            GENERAL ERROR ROUTINE             Y00.588022675ERRO1000
0043*  $      DKEND                                                                 ERRO1004
0044*  $      OBJECT                                              Y21.346031275PLS10000
0045*  $      DKEND                                                                 PLS10007
0046*+$       EXECUTE
0047*  $      FILE     10,X10S,15L
0048*  $      FILE     11,X11S,15L
0049*  $      LIMITS   15,23K
0050*  $      OPTION   FORTRAN,NOMAP
0051*  $      OBJECT            PASS 3 MAIN PROGRAM               Y21.265031275PASS3000
0052*  $      DKEND                                                                 PASS3035
0053*  $      OBJECT            GENERAL ERROR ROUTINE             Y00.472022675ERRO1000
0054*  $      DKEND                                                                 ERRO1004
0055*  $      OBJECT                                              Y21.437031275SAMOUT00
0056*  $      DKEND                                                                 SAMOUT18
0057*  $      OBJECT            FORTRAN UNIT GENERATOR ROUTINE    Y21.434031275FORS3000
```

```
0058* $      DKEND                                                          FORS3060
0059* $      OBJECT          FUNCTION GENERATOR 1                Y21.269031275GEN10000
0060* $      DKEND                                                          GEN10007
0061* $      OBJECT          FUNCTION GENERATOR 2                Y21.155031275GEN20000
0062* $      DKEND                                                          GEN20013
0063* $      OBJECT          FUNCTION GENERATOR 3                Y21.349031275GEN30000
0064* $      DKEND                                                          GEN30011
0065* $      OBJECT          PASS 3 DATA INPUTING ROUTINE        Y21.429031275DATA3000
0066* $      DKEND                                                          DATA3004
0067* $      OBJECT          CONTROL DATA SPECIFICATION FOR P Y00.461022675PARM0000
0068* $      DKEND                                                          PARM0003
0069*+$      EXECUTE
0070* $      FILE     11,X11S,15L
0071* $      FFILE    26,FIXLNG/1341,BUFSIZ/1344,NOSRLS,NSTDLB
0072  $      TAPE7    26,A11S,,20145,,SMIOSS,OUT                            00002770
0073  $      LIMITS   50,30K                                                00002780
0074  $      ENDJOB                                                         00002790
  TOTAL CARD COUNT THIS JOB = 000661


* ACTY-01  $CARD #0025* GELOAD   C9/11/77   SW=010000C00000
  * NORMAL TERMINATION      AT 015473 I=5020 SW=010000C00000

     START 14.489      LINES    298      PROC  0.0021      I/O   0.001     IU  1   MEMORY    13K
     STOP  14.493      LIMIT   5000      LIMIT 0.150C      LIMIT           CU  1   M*T       235
     SWAP   C.CCO
     LAPSE  C.004 FC D   TYPE      BUSY     IP/AT     FP/RT     IS/#C MS/#E     ADDRESS T#/PK#

                  I* R   D450 *    209        0        13        13    13     0-08-06
                  R* R   D450 *    137        0         0        11    11     0-08-06
                  10 S   D450 *    117        0         7       180   180     0-08-03
                  P*     SYOUT
                  L* R   D450 *   1590        0         0      1500  1500R    0-08-02
                  *L R   D450 *     39        0         0       400   400R    0-08-01

      LIST      24 LINES
      RC-52    274 LINES


     ACTIVITY COST:    $     .78        SERVICE LEVEL:  D
          D450: $      .09   PROC: $     .21   GEIN: $     .16   SYPR: $     .10
          CORE: $      .40   JHCB: $     .05


* ACTY-02  $CARD #0046* GELOAD   C9/11/77   SW=010000000000
  * NORMAL TERMINATION      AT 044163 I=5020 SW=010000000000

     START 14.494      LINES    305      PROC  0.0008      I/O   0.001     IU  1   MEMORY    23K
     STOP  14.495      LIMIT   5000      LIMIT 0.150C      LIMIT           CU  1   M*T       150
     SWAP   0.000
     LAPSE  0.001 FC D   TYPE      BUSY     IP/AT     FP/RT     IS/#C MS/#E     ADDRESS T#/PK#

                  10 S   D450 *     87        7         7       180   180     0-08-03
                  R* R   D450 *     94        0         0         7     7     0-08-06
                  11 S   D450 *     91        0         7       180   180     0-08-03
                  P*     SYOUT
                  L* R   D450 *    884        0         0      1500  1500R    0-08-02
                  *L R   D450 *     33        0         0       400   400R    0-08-01

      LIST      27 LINES
```

SNUMB = x2669, ACTIVIT # = 01, , REPORT CODE = 52, RECORD COUNT = 000274

```
SIA 0 4 175C0;
 INS 0 1;
 OSC P5 P6 B2 F1 P30;
 OUT B2 B1;
 END;
 GEN 0 2 1 1 1;
  NOT   2.0 1 2 119.523  14.60;
  NOT   2.0 1 2 119.523  29.20;
  NOT   2.0 1 2 119.523  43.80;
  NOT   2.0 1 2 119.523  58.40;
  NOT   2.0 1 2 119.523  73.00;
  NOT   2.0 1 2 119.523  87.60;
  NOT   2.0 1 2 119.523 102.20;
  NOT   5.0 1 2 104.654  14.60;
  NOT   5.0 1 2 104.654  29.20;
  NOT   5.0 1 2 104.654  43.80;
  NOT   5.0 1 2 104.654  58.40;
  NOT   5.0 1 2 185.164  73.00;
  NOT   5.0 1 2 104.654  87.60;
  NOT   5.0 1 2 104.654 102.20;
  NOT  11.0 1 2  98.071  14.60;
  NOT  11.0 1 2  98.071  29.20;
  NOT  11.0 1 2  98.071  43.80;
  NOT  11.0 1 2  98.071  58.40;
  NOT  11.0 1 2 205.635  73.00;
  NOT  11.0 1 2  98.071  87.60;
  NOT  11.0 1 2  98.071 102.20;
  NOT  14.0 1 2 119.523  14.60;
  NOT  14.0 1 2 119.523  29.20;
  NOT  14.0 1 2 119.523  43.80;
  NOT  14.0 1 2 119.523  58.40;
  NOT  14.0 1 2 119.523  73.00;
  NOT  14.0 1 2 119.523  87.60;
  NOT  14.0 1 2 119.523 102.20;
  NOT  20.0 1 2 119.523  14.60;
  NOT  20.0 1 2 119.523  29.20;
  NOT  20.0 1 2 119.523  43.80;
  NOT  20.0 1 2 119.523  58.40;
  NOT  20.0 1 2 119.523  73.00;
  NOT  20.0 1 2 119.523  87.60;
  NOT  20.0 1 2 119.523 102.20;
  NOT  23.0 1 2 127.354  14.60;
  NOT  23.0 1 2 127.354  29.20;
  NOT  23.0 1 2 127.354  43.80;
  NOT  23.0 1 2 127.354  58.40;
  NOT  23.0 1 2  51.824  73.00;
  NOT  23.0 1 2 127.354  87.60;
  NOT  23.0 1 2 127.354 102.20;
  NOT  29.0 1 2 110.841  14.60;
  NOT  29.0 1 2 110.841  29.20;
  NOT  29.0 1 2 110.841  43.80;
  NOT  29.0 1 2 110.841  58.40;
  NOT  29.0 1 2 162.129  73.00;
  NOT  29.0 1 2 110.841  87.60;
  NOT  29.0 1 2 110.841 102.20;
  NOT  32.0 1 2 119.523  14.60;
  NOT  32.0 1 2 119.523  29.20;
  NOT  32.0 1 2 119.523  43.80;
```

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NOT  32.0 1 2 119.523  58.40;
NOT  32.0 1 2 119.523  73.00;
NOT  32.0 1 2 119.523  87.60;
NOT  32.0 1 2 119.523 102.20;
NOT  38.0 1 2 119.523  14.60;
NOT  38.0 1 2 119.523  29.20;
NOT  38.0 1 2 119.523  43.80;
NOT  38.0 1 2 119.523  58.40;
NOT  38.0 1 2 119.523  73.00;
NOT  38.0 1 2 119.523  87.60;
NOT  38.0 1 2 119.523 102.20;
NOT  41.0 1 2 126.501  14.60;
NOT  41.0 1 2 126.501  29.20;
NOT  41.0 1 2 126.501  43.80;
NOT  41.0 1 2 126.501  58.40;
NOT  41.0 1 2  63.132  73.00;
NOT  41.0 1 2 126.501  87.60;
NOT  41.0 1 2 126.501 102.20;
NOT  47.0 1 2 129.044  14.60;
NOT  47.0 1 2 129.044  29.20;
NOT  47.0 1 2 129.044  43.80;
NOT  47.0 1 2 129.044  58.40;
NOT  47.0 1 2   9.258  73.00;
NOT  47.0 1 2 129.044  87.60;
NOT  47.0 1 2 129.044 102.20;
NOT  50.0 1 2 119.523  14.60;
NOT  50.0 1 2 119.523  29.20;
NOT  50.0 1 2 119.523  43.80;
NOT  50.0 1 2 119.523  58.40;
NOT  50.0 1 2 119.523  73.00;
NOT  50.0 1 2 119.523  87.60;
NOT  50.0 1 2 119.523 102.20;
NOT  56.0 1 2 119.523  14.60;
NOT  56.0 1 2 119.523  29.20;
NOT  56.0 1 2 119.523  43.80;
NOT  56.0 1 2 119.523  58.40;
NOT  56.0 1 2 119.523  73.00;
NOT  56.0 1 2 119.523  87.60;
NOT  56.0 1 2 119.523 102.20;
NOT  59.0 1 2 101.418  14.60;
NOT  59.0 1 2 101.418  29.20;
NOT  59.0 1 2 101.418  43.80;
NOT  59.0 1 2 101.418  58.40;
NOT  59.0 1 2 195.667  73.00;
NOT  59.0 1 2 101.418  87.60;
NOT  59.0 1 2 101.418 102.20;
NOT  65.0 1 2  94.617  14.60;
NOT  65.0 1 2  94.617  29.20;
NOT  65.0 1 2  94.617  43.80;
NOT  65.0 1 2  94.617  58.40;
NOT  65.0 1 2 215.141  73.00;
NOT  65.0 1 2  94.617  87.60;
NOT  65.0 1 2  94.617 102.20;
NOT  68.0 1 2 119.523  14.60;
NOT  68.0 1 2 119.523  29.20;
NOT  68.0 1 2 119.523  43.80;
NOT  68.0 1 2 119.523  58.40;
NOT  68.0 1 2 119.523  73.00;
NOT  68.0 1 2 119.523  87.60;
NOT  68.0 1 2 119.523 102.20;
```

```
NOT  74.0 1 2 126.072  14.60;
NOT  74.0 1 2 126.072  29.20;
NOT  74.0 1 2 126.072  43.80;
NOT  74.0 1 2 126.072  58.40;
NOT  74.0 1 2  68.086  73.00;
NOT  74.0 1 2 126.072  87.60;
NOT  74.0 1 2 126.072 102.20;
NOT  77.0 1 2 119.523  14.60;
NOT  77.0 1 2 119.523  29.20;
NOT  77.0 1 2 119.523  43.80;
NOT  77.0 1 2 119.523  58.40;
NOT  77.0 1 2 119.523  73.00;
NOT  77.0 1 2 119.523  87.60;
NOT  77.0 1 2 119.523 102.20;
NOT  83.0 1 2 119.523  14.60;
NOT  83.0 1 2 119.523  29.20;
NOT  83.0 1 2 119.523  43.80;
NOT  83.0 1 2 119.523  58.40;
NOT  83.0 1 2 119.523  73.00;
NOT  83.0 1 2 119.523  87.60;
NOT  83.0 1 2 119.523 102.20;
NOT  86.0 1 2 119.523  14.60;
NOT  86.0 1 2 119.523  29.20;
NOT  86.0 1 2 119.523  43.80;
NOT  86.0 1 2 119.523  58.40;
NOT  86.0 1 2 119.523  73.00;
NOT  86.0 1 2 119.523  87.60;
NOT  86.0 1 2 119.523 102.20;
NOT  92.0 1 2 119.523  14.60;
NOT  92.0 1 2 119.523  29.20;
NOT  92.0 1 2 119.523  43.80;
NOT  92.0 1 2 119.523  58.40;
NOT  92.0 1 2 119.523  73.00;
NOT  92.0 1 2 119.523  87.60;
NOT  92.0 1 2 119.523 102.20;
NOT  95.0 1 2 125.641  14.60;
NOT  95.0 1 2 125.641  29.20;
NOT  95.0 1 2 125.641  43.80;
NOT  95.0 1 2 125.641  58.40;
NOT  95.0 1 2  72.703  73.00;
NOT  95.0 1 2 125.641  87.60;
NOT  95.0 1 2 125.641 102.20;
NOT 101.0 1 2 119.523  14.60;
NOT 101.0 1 2 119.523  29.20;
NOT 101.0 1 2 119.523  43.80;
NOT 101.0 1 2 119.523  58.40;
NOT 101.0 1 2 119.523  73.00;
NOT 101.0 1 2 119.523  87.60;
NOT 101.0 1 2 119.523 102.20;
NOT 104.0 1 2 128.202  14.60;
NOT 104.0 1 2 128.202  29.20;
NOT 104.0 1 2 128.202  43.80;
NOT 104.0 1 2 128.202  58.40;
NOT 104.0 1 2  37.225  73.00;
NOT 104.0 1 2 128.202  87.60;
NOT 104.0 1 2 128.202 102.20;
NOT 110.0 1 2 119.523  14.60;
NOT 110.0 1 2 119.523  29.20;
NOT 110.0 1 2 119.523  43.80;
NOT 110.0 1 2 119.523  58.40;
```

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NOT 110.0 1 2 119.523  73.00;
NOT 110.0 1 2 119.523  87.60;
NOT 110.0 1 2 119.523 102.20;
NOT 113.0 1 2 113.808  14.60;
NOT 113.0 1 2 113.808  29.20;
NOT 113.0 1 2 113.808  43.80;
NOT 113.0 1 2 113.808  58.40;
NOT 113.0 1 2 149.284  73.00;
NOT 113.0 1 2 113.808  87.60;
NOT 113.0 1 2 113.808 102.20;
NOT 119.0 1 2 119.523  14.60;
NOT 119.0 1 2 119.523  29.20;
NOT 119.0 1 2 119.523  43.80;
NOT 119.0 1 2 119.523  58.40;
NOT 119.0 1 2 119.523  73.00;
NOT 119.0 1 2 119.523  87.60;
NOT 119.0 1 2 119.523 102.20;
NOT 122.0 1 2 127.779  14.60;
NOT 122.0 1 2 127.779  29.20;
NOT 122.0 1 2 127.779  43.80;
NOT 122.0 1 2 127.779  58.40;
NOT 122.0 1 2  45.119  73.00;
NOT 122.0 1 2 127.779  87.60;
NOT 122.0 1 2 127.779 102.20;
NOT 128.0 1 2 119.523  14.60;
NOT 128.0 1 2 119.523  29.20;
NOT 128.0 1 2 119.523  43.80;
NOT 128.0 1 2 119.523  58.40;
NOT 128.0 1 2 119.523  73.00;
NOT 128.0 1 2 119.523  87.60;
NOT 128.0 1 2 119.523 102.20;
NOT 131.0 1 2 126.928  14.60;
NOT 131.0 1 2 126.928  29.20;
NOT 131.0 1 2 126.928  43.80;
NOT 131.0 1 2 126.928  58.40;
NOT 131.0 1 2  57.756  73.00;
NOT 131.0 1 2 126.928  87.60;
NOT 131.0 1 2 126.928 102.20;
NOT 137.0 1 2  91.026  14.60;
NOT 137.0 1 2  91.026  29.20;
NOT 137.0 1 2  91.026  43.80;
NOT 137.0 1 2  91.026  58.40;
NOT 137.0 1 2 224.245  73.00;
NOT 137.0 1 2  91.026  87.60;
NOT 137.0 1 2  91.026 102.20;
NOT 140.0 1 2 119.523  14.60;
NOT 140.0 1 2 119.523  29.20;
NOT 140.0 1 2 119.523  43.80;
NOT 140.0 1 2 119.523  58.40;
NOT 140.0 1 2 119.523  73.00;
NOT 140.0 1 2 119.523  87.60;
NOT 140.0 1 2 119.523 102.20;
NOT 146.0 1 2 107.792  14.60;
NOT 146.0 1 2 107.792  29.20;
NOT 146.0 1 2 107.792  43.80;
NOT 146.0 1 2 107.792  58.40;
NOT 146.0 1 2 174.028  73.00;
NOT 146.0 1 2 107.792  87.60;
NOT 146.0 1 2 107.792 102.20;
NOT 149.0 1 2 119.523  14.60;
```

```
NOT 149.0 1 2 119.523  29.20;
NOT 149.0 1 2 119.523  43.80;
NOT 149.0 1 2 119.523  58.40;
NOT 149.0 1 2 119.523  73.00;
NOT 149.0 1 2 119.523  87.60;
NOT 149.0 1 2 119.523 102.20;
NOT 155.0 1 2 119.523  14.60;
NOT 155.0 1 2 119.523  29.20;
NOT 155.0 1 2 119.523  43.80;
NOT 155.0 1 2 119.523  58.40;
NOT 155.0 1 2 119.523  73.00;
NOT 155.0 1 2 119.523  87.60;
NOT 155.0 1 2 119.523 102.20;
NOT 158.0 1 2 116.701  14.60;
NOT 158.0 1 2 116.701  29.20;
NOT 158.0 1 2 116.701  43.80;
NOT 158.0 1 2 116.701  58.40;
NOT 158.0 1 2 135.225  73.00;
NOT 158.0 1 2 116.701  87.60;
NOT 158.0 1 2 116.701 102.20;
NOT 164.0 1 2 128.624  14.60;
NOT 164.0 1 2 128.624  29.20;
NOT 164.0 1 2 128.624  43.80;
NOT 164.0 1 2 128.624  58.40;
NOT 164.0 1 2  27.124  73.00;
NOT 164.0 1 2 128.624  87.60;
NOT 164.0 1 2 128.624 102.20;
NOT 167.0 1 2 119.523  14.60;
NOT 167.0 1 2 119.523  29.20;
NOT 167.0 1 2 119.523  43.80;
NOT 167.0 1 2 119.523  58.40;
NOT 167.0 1 2 119.523  73.00;
NOT 167.0 1 2 119.523  87.60;
NOT 167.0 1 2 119.523 102.20;
TERM 169.0;
END OF PASS I
```

PASS II REPORT

(WORD CNT)								
4	12.000	0.	4.000	17500.000				
3	2.000	0.	1.000					
8	2.000	0.	102.000	5.000	6.000	-2.000	-101.000	30.000
5	2.000	0.	101.000	-2.000	-1.000			
2	2.000	0.						
6	3.000	0.	2.000	1.000	1.000	1.000		
6	1.000	2.000	1.000	2.000	119.523	14.600		
6	1.000	2.000	1.000	2.000	119.523	29.200		
6	1.000	2.000	1.000	2.000	119.523	43.800		
6	1.000	2.000	1.000	2.000	119.523	58.400		
6	1.000	2.000	1.000	2.000	119.523	73.000		
6	1.000	2.000	1.000	2.000	119.523	87.600		
6	1.000	2.000	1.000	2.000	119.523	102.200		
6	1.000	5.000	1.000	2.000	104.654	14.600		
6	1.000	5.000	1.000	2.000	104.654	29.200		
6	1.000	5.000	1.000	2.000	104.654	43.800		
6	1.000	5.000	1.000	2.000	104.654	58.400		
6	1.000	5.000	1.000	2.000	185.164	73.000		
6	1.000	5.000	1.000	2.000	104.654	87.600		
6	1.000	5.000	1.000	2.000	104.654	102.200		
6	1.000	11.000	1.000	2.000	98.071	14.600		
6	1.000	11.000	1.000	2.000	98.071	29.200		
6	1.000	11.000	1.000	2.000	98.071	43.800		
6	1.000	11.000	1.000	2.000	98.071	58.400		
6	1.000	11.000	1.000	2.000	205.635	73.000		
6	1.000	11.000	1.000	2.000	98.071	87.600		
6	1.000	11.000	1.000	2.000	98.071	102.200		
6	1.000	14.000	1.000	2.000	119.523	14.600		
6	1.000	14.000	1.000	2.000	119.523	29.200		
6	1.000	14.000	1.000	2.000	119.523	43.800		
6	1.000	14.000	1.000	2.000	119.523	58.400		
6	1.000	14.000	1.000	2.000	119.523	73.000		
6	1.000	14.000	1.000	2.000	119.523	87.600		
6	1.000	14.000	1.000	2.000	119.523	102.200		
6	1.000	20.000	1.000	2.000	119.523	14.600		
6	1.000	20.000	1.000	2.000	119.523	29.200		
6	1.000	20.000	1.000	2.000	119.523	43.800		
6	1.000	20.000	1.000	2.000	119.523	58.400		
6	1.000	20.000	1.000	2.000	119.523	73.000		
6	1.000	20.000	1.000	2.000	119.523	87.600		
6	1.000	20.000	1.000	2.000	119.523	102.200		
6	1.000	23.000	1.000	2.000	127.354	14.600		
6	1.000	23.000	1.000	2.000	127.354	29.200		
6	1.000	23.000	1.000	2.000	127.354	43.800		
6	1.000	23.000	1.000	2.000	127.354	58.400		
6	1.000	23.000	1.000	2.000	51.824	73.000		
6	1.000	23.000	1.000	2.000	127.354	87.600		
6	1.000	23.000	1.000	2.000	127.354	102.200		
6	1.000	29.000	1.000	2.000	110.841	14.600		
6	1.000	29.000	1.000	2.000	110.841	29.200		
6	1.000	29.000	1.000	2.000	110.841	43.800		
6	1.000	29.000	1.000	2.000	110.841	58.400		
6	1.000	29.000	1.000	2.000	162.129	73.000		
6	1.000	29.000	1.000	2.000	110.841	87.600		
6	1.000	29.000	1.000	2.000	110.841	102.200		
6	1.000	32.000	1.000	2.000	119.523	14.600		
6	1.000	32.000	1.000	2.000	119.523	29.200		

6	1.000	32.000	1.000	2.000	119.523	43.800
6	1.000	32.000	1.000	2.000	119.523	58.400
6	1.000	32.000	1.000	2.000	119.523	73.000
6	1.000	32.000	1.000	2.000	119.523	87.600
6	1.000	32.000	1.000	2.000	119.523	102.200
6	1.000	38.000	1.000	2.000	119.523	14.600
6	1.000	38.000	1.000	2.000	119.523	29.200
6	1.000	38.000	1.000	2.000	119.523	43.800
6	1.000	38.000	1.000	2.000	119.523	58.400
6	1.000	38.000	1.000	2.000	119.523	73.000
6	1.000	38.000	1.000	2.000	119.523	87.600
6	1.000	38.000	1.000	2.000	119.523	102.200
6	1.000	41.000	1.000	2.000	126.501	14.600
6	1.000	41.000	1.000	2.000	126.501	29.200
6	1.000	41.000	1.000	2.000	126.501	43.800
6	1.000	41.000	1.000	2.000	126.501	58.400
6	1.000	41.000	1.000	2.000	63.132	73.000
6	1.000	41.000	1.000	2.000	126.501	87.600
6	1.000	41.000	1.000	2.000	126.501	102.200
6	1.000	47.000	1.000	2.000	129.044	14.600
6	1.000	47.000	1.000	2.000	129.044	29.200
6	1.000	47.000	1.000	2.000	129.044	43.800
6	1.000	47.000	1.000	2.000	129.044	58.400
6	1.000	47.000	1.000	2.000	9.258	73.000
6	1.000	47.000	1.000	2.000	129.044	87.600
6	1.000	47.000	1.000	2.000	129.044	102.200
6	1.000	50.000	1.000	2.000	119.523	14.600
6	1.000	50.000	1.000	2.000	119.523	29.200
6	1.000	50.000	1.000	2.000	119.523	43.800
6	1.000	50.000	1.000	2.000	119.523	58.400
6	1.000	50.000	1.000	2.000	119.523	73.000
6	1.000	50.000	1.000	2.000	119.523	87.600
6	1.000	50.000	1.000	2.000	119.523	102.200
6	1.000	56.000	1.000	2.000	119.523	14.600
6	1.000	56.000	1.000	2.000	119.523	29.200
6	1.000	56.000	1.000	2.000	119.523	43.800
6	1.000	56.000	1.000	2.000	119.523	58.400
6	1.000	56.000	1.000	2.000	119.523	73.000
6	1.000	56.000	1.000	2.000	119.523	87.600
6	1.000	56.000	1.000	2.000	119.523	102.200
6	1.000	59.000	1.000	2.000	101.418	14.600
6	1.000	59.000	1.000	2.000	101.418	29.200
6	1.000	59.000	1.000	2.000	101.418	43.800
6	1.000	59.000	1.000	2.000	101.418	58.400
6	1.000	59.000	1.000	2.000	195.667	73.000
6	1.000	59.000	1.000	2.000	101.418	87.600
6	1.000	59.000	1.000	2.000	101.418	102.200
6	1.000	65.000	1.000	2.000	94.617	14.600
6	1.000	65.000	1.000	2.000	94.617	29.200
6	1.000	65.000	1.000	2.000	94.617	43.800
6	1.000	65.000	1.000	2.000	94.617	58.400
6	1.000	65.000	1.000	2.000	215.141	73.000
6	1.000	65.000	1.000	2.000	94.617	87.600
6	1.000	65.000	1.000	2.000	94.617	102.200
6	1.000	68.000	1.000	2.000	119.523	14.600
6	1.000	68.000	1.000	2.000	119.523	29.200
6	1.000	68.000	1.000	2.000	119.523	43.800
6	1.000	68.000	1.000	2.000	119.523	58.400
6	1.000	68.000	1.000	2.000	119.523	73.000
6	1.000	68.000	1.000	2.000	119.523	87.600

6	1.000	68.000	1.000	2.000	119.523	102.200
6	1.000	74.000	1.000	2.000	126.072	14.600
6	1.000	74.000	1.000	2.000	126.072	29.200
6	1.000	74.000	1.000	2.000	126.072	43.800
6	1.000	74.000	1.000	2.000	126.072	58.400
6	1.000	74.000	1.000	2.000	68.086	73.000
6	1.000	74.000	1.000	2.000	126.072	87.600
6	1.000	74.000	1.000	2.000	126.072	102.200
6	1.000	77.000	1.000	2.000	119.523	14.600
6	1.000	77.000	1.000	2.000	119.523	29.200
6	1.000	77.000	1.000	2.000	119.523	43.800
6	1.000	77.000	1.000	2.000	119.523	58.400
6	1.000	77.000	1.000	2.000	119.523	73.000
6	1.000	77.000	1.000	2.000	119.523	87.600
6	1.000	77.000	1.000	2.000	119.523	102.200
6	1.000	83.000	1.000	2.000	119.523	14.600
6	1.000	83.000	1.000	2.000	119.523	29.200
6	1.000	83.000	1.000	2.000	119.523	43.800
6	1.000	83.000	1.000	2.000	119.523	58.400
6	1.000	83.000	1.000	2.000	119.523	73.000
6	1.000	83.000	1.000	2.000	119.523	87.600
6	1.000	83.000	1.000	2.000	119.523	102.200
6	1.000	86.000	1.000	2.000	119.523	14.600
6	1.000	86.000	1.000	2.000	119.523	29.200
6	1.000	86.000	1.000	2.000	119.523	43.800
6	1.000	86.000	1.000	2.000	119.523	58.400
6	1.000	86.000	1.000	2.000	119.523	73.000
6	1.000	86.000	1.000	2.000	119.523	87.600
6	1.000	86.000	1.000	2.000	119.523	102.200
6	1.000	92.000	1.000	2.000	119.523	14.600
6	1.000	92.000	1.000	2.000	119.523	29.200
6	1.000	92.000	1.000	2.000	119.523	43.800
6	1.000	92.000	1.000	2.000	119.523	58.400
6	1.000	92.000	1.000	2.000	119.523	73.000
6	1.000	92.000	1.000	2.000	119.523	87.600
6	1.000	92.000	1.000	2.000	119.523	102.200
6	1.000	95.000	1.000	2.000	125.641	14.600
6	1.000	95.000	1.000	2.000	125.641	29.200
6	1.000	95.000	1.000	2.000	125.641	43.800
6	1.000	95.000	1.000	2.000	125.641	58.400
6	1.000	95.000	1.000	2.000	72.703	73.000
6	1.000	95.000	1.000	2.000	125.641	87.600
6	1.000	95.000	1.000	2.000	125.641	102.200
6	1.000	101.000	1.000	2.000	119.523	14.600
6	1.000	101.000	1.000	2.000	119.523	29.200
6	1.000	101.000	1.000	2.000	119.523	43.800
6	1.000	101.000	1.000	2.000	119.523	58.400
6	1.000	101.000	1.000	2.000	119.523	73.000
6	1.000	101.000	1.000	2.000	119.523	87.600
6	1.000	101.000	1.000	2.000	119.523	102.200
6	1.000	104.000	1.000	2.000	128.202	14.600
6	1.000	104.000	1.000	2.000	128.202	29.200
6	1.000	104.000	1.000	2.000	128.202	43.800
6	1.000	104.000	1.000	2.000	128.202	58.400
6	1.000	104.000	1.000	2.000	37.825	73.000
6	1.000	104.000	1.000	2.000	128.202	87.600
6	1.000	104.000	1.000	2.000	128.202	102.200
6	1.000	110.000	1.000	2.000	119.523	14.600
6	1.000	110.000	1.000	2.000	119.523	29.200
6	1.000	110.000	1.000	2.000	119.523	43.800

6	1.000	110.000	1.000	2.000	119.523	58.400
6	1.000	110.000	1.000	2.000	119.523	73.000
6	1.000	110.000	1.000	2.000	119.523	87.600
6	1.000	110.000	1.000	2.000	119.523	102.200
6	1.000	113.000	1.000	2.000	113.808	14.600
6	1.000	113.000	1.000	2.000	113.808	29.200
6	1.000	113.000	1.000	2.000	113.808	43.800
6	1.000	113.000	1.000	2.000	113.808	58.400
6	1.000	113.000	1.000	2.000	149.284	73.000
6	1.000	113.000	1.000	2.000	113.808	87.600
6	1.000	113.000	1.000	2.000	113.808	102.200
6	1.000	119.000	1.000	2.000	119.523	14.600
6	1.000	119.000	1.000	2.000	119.523	29.200
6	1.000	119.000	1.000	2.000	119.523	43.800
6	1.000	119.000	1.000	2.000	119.523	58.400
6	1.000	119.000	1.000	2.000	119.523	73.000
6	1.000	119.000	1.000	2.000	119.523	87.600
6	1.000	119.000	1.000	2.000	119.523	102.200
6	1.000	122.000	1.000	2.000	127.779	14.600
6	1.000	122.000	1.000	2.000	127.779	29.200
6	1.000	122.000	1.000	2.000	127.779	43.800
6	1.000	122.000	1.000	2.000	127.779	58.400
6	1.000	122.000	1.000	2.000	45.119	73.000
6	1.000	122.000	1.000	2.000	127.779	87.600
6	1.000	122.000	1.000	2.000	127.779	102.200
6	1.000	128.000	1.000	2.000	119.523	14.600
6	1.000	128.000	1.000	2.000	119.523	29.200
6	1.000	128.000	1.000	2.000	119.523	43.800
6	1.000	128.000	1.000	2.000	119.523	58.400
6	1.000	128.000	1.000	2.000	119.523	73.000
6	1.000	128.000	1.000	2.000	119.523	87.600
6	1.000	128.000	1.000	2.000	119.523	102.200
6	1.000	131.000	1.000	2.000	126.928	14.600
6	1.000	131.000	1.000	2.000	126.928	29.200
6	1.000	131.000	1.000	2.000	126.928	43.800
6	1.000	131.000	1.000	2.000	126.928	58.400
6	1.000	131.000	1.000	2.000	57.756	73.000
6	1.000	131.000	1.000	2.000	126.928	87.600
6	1.000	131.000	1.000	2.000	126.928	102.200
6	1.000	137.000	1.000	2.000	91.026	14.600
6	1.000	137.000	1.000	2.000	91.026	29.200
6	1.000	137.000	1.000	2.000	91.026	43.800
6	1.000	137.000	1.000	2.000	91.026	58.400
6	1.000	137.000	1.000	2.000	224.245	73.000
6	1.000	137.000	1.000	2.000	91.026	87.600
6	1.000	137.000	1.000	2.000	91.026	102.200
6	1.000	140.000	1.000	2.000	119.523	14.600
6	1.000	140.000	1.000	2.000	119.523	29.200
6	1.000	140.000	1.000	2.000	119.523	43.800
6	1.000	140.000	1.000	2.000	119.523	58.400
6	1.000	140.000	1.000	2.000	119.523	73.000
6	1.000	140.000	1.000	2.000	119.523	87.600
6	1.000	140.000	1.000	2.000	119.523	102.200
6	1.000	146.000	1.000	2.000	107.792	14.600
6	1.000	146.000	1.000	2.000	107.792	29.200
6	1.000	146.000	1.000	2.000	107.792	43.800
6	1.000	146.000	1.000	2.000	107.792	58.400
6	1.000	146.000	1.000	2.000	174.028	73.000
6	1.000	146.000	1.000	2.000	107.792	87.600
6	1.000	146.000	1.000	2.000	107.792	102.200

6	1.000	149.000	1.000	2.000	119.523	14.600
6	1.000	149.000	1.000	2.000	119.523	29.200
6	1.000	149.000	1.000	2.000	119.523	43.800
6	1.000	149.000	1.000	2.000	119.523	58.400
6	1.000	149.000	1.000	2.000	119.523	73.000
6	1.000	149.000	1.000	2.000	119.523	87.600
6	1.000	149.000	1.000	2.000	119.523	102.200
6	1.000	155.000	1.000	2.000	119.523	14.600
6	1.000	155.000	1.000	2.000	119.523	29.200
6	1.000	155.000	1.000	2.000	119.523	43.800
6	1.000	155.000	1.000	2.000	119.523	58.400
6	1.000	155.000	1.000	2.000	119.523	73.000
6	1.000	155.000	1.000	2.000	119.523	87.600
6	1.000	155.000	1.000	2.000	119.523	102.200
6	1.000	158.000	1.000	2.000	116.701	14.600
6	1.000	158.000	1.000	2.000	116.701	29.200
6	1.000	158.000	1.000	2.000	116.701	43.800
6	1.000	158.000	1.000	2.000	116.701	58.400
6	1.000	158.000	1.000	2.000	135.225	73.000
6	1.000	158.000	1.000	2.000	116.701	87.600
6	1.000	158.000	1.000	2.000	116.701	102.200
6	1.000	164.000	1.000	2.000	128.624	14.600
6	1.000	164.000	1.000	2.000	128.624	29.200
6	1.000	164.000	1.000	2.000	128.624	43.800
6	1.000	164.000	1.000	2.000	128.624	58.400
6	1.000	164.000	1.000	2.000	27.124	73.000
6	1.000	164.000	1.000	2.000	128.624	87.600
6	1.000	164.000	1.000	2.000	128.624	102.200
6	1.000	167.000	1.000	2.000	119.523	14.600
6	1.000	167.000	1.000	2.000	119.523	29.200
6	1.000	167.000	1.000	2.000	119.523	43.800
6	1.000	167.000	1.000	2.000	119.523	58.400
6	1.000	167.000	1.000	2.000	119.523	73.000
6	1.000	167.000	1.000	2.000	119.523	87.600
6	1.000	167.000	1.000	2.000	119.523	102.200
2	6.000	169.000				

END OF PASS II

APPENDIX C

Card Image Generation Program

```
0010 INTEGER Z(5)/80252,80254,20145,20127,23176/
0015 CHARACTER A*6(5)/'PSYCH0','PSYCH1','SMIOS5','SMIOS1',
0016&'CP+CSH'/
80 WRITE(6,1)
90 1 FORMAT(////)
100 DIMENSION PC(19),PO(19),F(7),IR(19)
110 PRINT,"ENTER PARTIAL TO BE ALTERED"
120 READ(5,10)IPC
130 10 FORMAT(I1)
140 PRINT,"ENTER 19 LEVELS OF PS, GREATEST TO SMALLEST"
150 READ(20,20)(PC(J),J=1,19)
160 20 FORMAT(5X,F7.3)
170 PRINT,"ENTER 19 LEVELS OF PC, SMALLEST TO GREATEST"
180 READ(20,20)(PO(J1),J1=1,19)
190 PRINT,"ENTER FREQUENCY PARAMETERS 1-7:"
200 READ(20,40)(F(J2),J2=1,7)
210 40 FORMAT(5X,F5.2)
220 PRINT,"ENTER 1 OR 0 FOR 19 TRIALS, SPECIFYING STANDARD
230& OR COMPARISON 1ST PER PAIR:"
240 READ(20,50)(IR(J3),J3=1,19)
250 50 FORMAT(5X,I1)
260 WRITE(21,200)
270 200 FORMAT('$:IDENT:5919KEN$MUSICFILE*D,R20GER',/,'$:SELECT:
280&5919KEN/MUSIC1',/,'SIA 0 4 17500;',/,'INS 0 1;',/,
290&'OSC P5 P6 B2 F1 P30;',/,'OUT B2 B1;',/,'END;',/,'GEN 0 2 1
300& 1 1;')
310 T=-1.0
320 DO 55 K=1,19
330 T=T+3.0
340 IF(IR(K).EQ.0)GO TO 70
350 59 DO 65 K1=1,7
360 P=PC(10)
370 WRITE(21,60) T,P,F(K1)
380 60 FORMAT( 1X,"NOT ",F5.1," 1 2 ",F7.3,1X,F6.2,";")
390 65 CONTINUE
400 T=T+3.0
410 IF(IR(K).EQ.0)GO TO 55
420 70 DO 75 K2=1,7
430 IF(K2.NE.IPC)GO TO 80
440 P=PC(K)
450 WRITE(21,60) T,P,F(K2)
460 GO TO 75
470 80 P=PO(K)
480 WRITE(21,60) T,P,F(K2)
490 75 CONTINUE
500 T=T+3.0
510 IF(IR(K).EQ.0)GO TO 59
520 55 CONTINUE
530 T=T-1.0
540 WRITE(21,201)T
550 201 FORMAT(1X,'TERM ',F5.1,';')
560 WRITE(21,1101)
570 1101 FORMAT("$:SELECT:5919KEN/MUSIC2")
580 PRINT,"ENTER TAPE FOR DATA, 1-5:"
590 READ(5,113)I4
600 113 FORMAT(I1)
630 WRITE(21,211)Z(I4),A(I4)
640 211 FORMAT('$:TAPE7:26,A11S,,',I5,',,',A6,',OUT')
650 WRITE(21,212)
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```
212 FORMAT('$:LIMITS:50,30K',/,'$:ENDJOB')
STOP
END
```

APPENDIX D

Demonstration and Data Collecting Programs

```
1 REM  DATA COLLECTION PROGRAM
2 REM  WRITTEN BY ROGER A. KENDA
LL, 1979
3 REM  TRS-80  MICROSOFT BASIC
5 CLEAR 1000
10 DIMA(19),S(3,19),R$(28,19),TI
(8),T(28,2)
30 DATA 67.01,66.65,66.26,65.83,
65.35,64.81,64.21,63.48,62.62,61
.55,57.23,56.66,56.01,55.23,54.2
9,53.09,51.42,48.67,39.33
35 DATA 5,3,15,7,13,19,4,2,12,18
,11,17,8,16,14,1,6,9,18
40 DATA16,17,5,3,2,12,19,4,6,10,
18,7,11,13,14,15,8,9,1
50 DATA 18,9,6,1,14,16,8,17,11,1
0,12,2,4,19,13,7,15,3,5
55 INPUT"WHAT IS THE SUBJECT'S N
AME";N$
60 FORN=1TO19:READA(N):NEXT
70 FORX=1TO3
80 FORN=1TO19
90 READS(X,N)
100 NEXT N,X
101 CLS
110 PRINTCHR$(23):PRINT"DIFFEREN
CE THRESHOLDS"
120 PRINT"FOR TIMBRE":PRINT:PRIN
T:PRINT"PRESS ENTER TO CONTINUE"

130 INPUTZ$:CLS
135 INPUT"DO YOU NEED DIRECTIONS
";A$:IFA$()"YES"THEN250
140 CLS:PRINT"BRIEF DIRECTIONS:"
:PRINT:PRINT
150 PRINT"YOU WILL BE PRESENTED
WITH 19 PAIRS OF TONES IN EACH S
ERIES"
155 PRINT"AFTER EACH PAIR, DECID
E WHETHER THE TONES HAD THE SAME
 OR "
160 PRINT"DIFFERENT TIMBRE --- I
F YOU ARE NOT SURE, RESPOND 'SAM
E'."
165 PRINT"RESPOND BY PRESSING TH
E 'S' KEY ON THE LEFT SIDE FOR S
AME"
170 PRINT"OR THE 'D' KEY ON THE
RIGHT SIDE OF THE KEYBOARD FOR D
IFFERENT."
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175 PRINT"IF YOU MAKE AN ERROR,
PRESS THE SPACE BAR AND CHANGE Y
OUR RESPONSE."
200 PRINT"THE COMPUTER WILL KEEP
 TRACK OF THE TRIAL NUMBER AND Y
OUR RESPONSES."
205 PRINT:PRINT:PRINT"CONCENTRAT
E ON JUDGING THE TIMBRE --- NOT
PITCH OR LOUDNESS."
210 INPUT"PRESS ENTER TO START P
ROGRAM";Z$
250 CLS
255 GOTO2000
256 FORL=1TO220:NEXT:CLS
257 PRINT@0,"SUBJECT ";M$
260 K=K+1:IFK=20THEN350ELSE PRIN
T@384,"RESPONSE FOR PAIR #";K;"2
";
270 GOSUB1000
280 IFA$="S"THEN PRINT@474,"SAME
 "
290 IFA$="K"THEN PRINT@474,"DIFF
EREMT                      ";A
$="D"
300 IFA$<>"S"AMD A$<>"K"THEN PRI
NT@474,"ERROR!!--RETYPE":GOTO270

310 R$(T,S(R,K))=A$
320 GOTO256
350 PRINT"END OF SERIES";T
360 GOTO3000
400 REM DATA PRESENTATION
405 INPUT"WHAT SERIES #";T1
409 CLS
410 FOR M=1TO15
420 PRINTM;A(M),R$(T1,M)
430 NEXT
431 INPUT "READY";Z$:FORM=16TO19
:PRINTM;A(M),R$(T1,M)
432 NEXT
445 PRINT"DO YOU WANT TO INDICAT
E THRESHOLD POINTS FOR SERIES";T
1
446 INPUTA$:IFA$<>"YES"GOTO3000
460 INPUT"INDICATE PAIR: UPPER T
HRESHOLD, LOWER THRESHOLD";U,L
470 T(T1,1)=U:T(T1,2)=L
475 GOTO3000
500 PRINT"INPUT TRIAL NUMBERS TO
 USE IN CALCULATIONS, NEGATIVE E
XITS"
510 FOR M=1TO6
520 INPUT T1(M):IFT1(M)<0THEN600
```

```
530 NEXT
600 M1=M-1
601 UL=0:LL=0
610 FORX=1TOM1
615 U1=(A(T(T1(X),1))+(A(T(T1(X)
,1)-1)))/2
620 L1=(A(T(T1(X),2))+(A(T(T1(X)
,2)+1)))/2
621 PRINT"FOR TRIAL";T1(X);"UPPE
R THRESHOLD =";U1;"LOWER THRESHO
LD = ";L1
622 UL=UL+U1:LL=LL+L1
625 NEXT
626 UL=UL/M1:LL=LL/M1
630 IU=UL-LL:DT=IU/2:PE=(UL+LL)/
2
650 PRINT"MEAN UPPER THRESHOLD =
";UL;" LOWER THRESHOLD =";LL
655 PRINT"INTERVAL OF UNCERTAINT
Y =";IU
660 PRINT "P.S.E. =";PE
665 PRINT"DIFFERENCE THRESHOLD =
";DT
670 GOTO3000
1000 A$=INKEY$:IFA$=""THEN1000
1010 IFA$<>CHR$(32)THEN RETURN
1015 K=K-1:PRINT@384,"CHANGE RES
PONSE FOR TRIAL #";K:PRINT@474,"
TYPE NEW RESPONSE FOR PAIR";K
1016 GOTO1000
2000 X=0
2020 INPUT"INPUT R.S.N";R
2021 IFR>3 OR R<1 THEN PRINT"ERR
OR!":GOTO2020
2030 INPUT"INPUT SERIES #";T
2040 GOTO256
3000 PRINT"MENU:    "
3001 PRINT"      1.  ADDITIONAL
SERIES"
3002 PRINT"      2.  RESPONSE PR
ESENTATION"
3003 PRINT"      3.  DIFFERENCE
THRESHOLD CALCULATIONS"
3005 PRINT"      4.  TERMINATE P
ROGRAM"
3006 INPUT"WHAT OPTION";V:ONVGOT
O255,400,500,4000
4000 PRINT"DONE":END
```

```
1 REM ORIENTATION PROGRAM
2 REM DEMONSTRATION OF TIMBRE PR
INCIPLES FOR SUBJECTS.
3 REM WRITTEN FOR TRS-80 MICROCO
MPUTER IN MICROSOFT BASIC.
4 REM 128 X 48 GRAPHICS REQUIRED

5 CLEAR 100:CLS:DIMA(128),A2(128
)
6 PRINTCHR$(23)
10 FOR N=1TO15:PRINTSTRING$(64,1
34);:NEXT
11 PRINT@342,"ORIENTATION ";:PRI
NT@406,"PRESENTATION ";:PRINT@47
0,"OM";:PRINT@534,"TIMBRE";
15 GOSUB3000
16 GOSUB4000
20 CLS:PRINT"THIS PRESENTATION W
ILL ACQUAINT YOU WITH THE PURPOS
E"
25 PRINT"OF THIS INVESTIGATION.
 IT IS VERY IMPORTANT THAT YOU U
NDERSTAND"
30 PRINT"WHAT TIMBRE IS SO THAT
WHEN YOU LISTEN FOR DIFFERENCES"

33 PRINT"IN TIMBRE YOU WILL MAKE
 ACCURATE JUDGEMENTS."
40 PRINT"AT THE END OF EACH PAGE
, PRESS ANY KEY TO CONTINUE"
45 GOSUB2000
50 PRINT"SOUND IS THE SENSATION
CAUSED BY VIBRATIONS IN THE MIDD
LE AND INNER EAR WHICH"
55 PRINT"HAVE BEEN TRANSMITTED T
O NERVE CELLS."
60 PRINT:PRINT:PRINT"THESE VIBRA
TIONS TRAVEL THROUGH ANY SUFFICI
ENTLY DENSE"
65 PRINT"MEDIUM, BUT ARE MOST CO
MMONLY TRANSFERED BY THE AIR."
70 PRINT:PRINT"THE SIMPLEST TYPE
 OF VIBRATION IS THAT OF A PENDU
LUM."
75 PRINT"IF A PENDULUM IS SWUNG
BACK AND FORTH, IT PRESCRIBES AN
 ARC."
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```
80 PRINT"IF A PIECE OF PAPER WER
E SLID UNDERNEATH THE PENDULUM"
85 PRINT"AT A UNIFORM RATE, AND
THE PENDULUM HAD A PEN POINT,"
86 PRINT"THE FOLLOWING FORM OF V
IBRATION WOULD APPEAR:":GOSUB200
0
95 IP=15
100 PRINT@18,"(PENDULUM AS SEEN
FROM THE TOP LOOKING DOWN)"
105 FOR W=76TO972STEP64:PRINT@W,
STRING$(51,191)::NEXT
110 H=15438:PRINT@179,"PAPER";:P
RINT@910,"PAPER";
115 FOR Y=23TO0STEP-2:SET(0,Y):S
ET(1,Y):GOSUB1500:RESET(0,Y):RES
ET(1,Y):NEXT
120 FORN=1TO2:FORY=0TO47STEP2:SE
T(0,Y):SET(1,Y):GOSUB1500:RESET(
0,Y):RESET(1,Y):NEXTY
125 FOR Y=47TO0STEP-2
130 SET(0,Y):SET(1,Y):GOSUB1500:
RESET(0,Y):RESET(1,Y):NEXTY
135 NEXTN:FORY=0TO47STEP2:SET(0,
Y):SET(1,Y):GOSUB1500:RESET(0,Y)
:RESET(1,Y):NEXT
140 FORY=47TO23STEP-2:J=J+1:H=H-
1:SET(0,Y):SET(1,Y):FORU=0TO14:P
OKE H+(U*64),191:NEXTU:RESET(0,Y
):RESET(1,Y):NEXTY
145 FORX=1TO120:FORU=1TO15:NEXTU
:RESET(X,A(X)):NEXTX
150 PRINT@645,"THIS PATTERN IS K
NOWN AS THE";:PRINT@709,"WAVEFOR
M OF THE VIBRATION.";
155 PRINT@773,"THIS SIMPLEST PAT
TERN IS";
160 PRINT@837,"CALLED A SINE CUR
VE";
165 PRINT@910,STRING$(5,191)::GO
SUB1000
170 PRINT@645,"THE CENTER OF THE
 SCREEN IS";:PRINT@709,"THE POIN
T OF EQUILIBRIUM";
175 PRINT@773,"THE PENDULUM IS I
N ITS":
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```
180 PRINT@837,"CENTER POSITION";

185 PRINT@18,"(DRAWING OF THE PO
INT OF EQUILIBRIUM)     ";
190 FOR N=0TO127:RESET(N,23):NEX
T:GOSUB1000
195 PRINT@645,"THE HEIGHT OF THE
 CURVE FROM";
200 PRINT@709,"EQUILIBRIUM IS KN
OWN AS THE";
205 PRINT@773,"AMPLITUDE";
210 FOR Y=8TO23:RESET(31,Y):NEXT
:PRINT@388,"AMPLITUDE";CHR$(94);

215 PRINT@18,"(DRAWING IN THE HI
GHEST AMPLITUDE POSITION)   ";
220 GOSUB1000
225 PRINT@645,"AMPLITUDE IS RELA
TED TO THE";
230 PRINT@709,"AMOUNT OF ENERGY
IN THE WAVE";
235 PRINT@773,"IT IS ALSO RELATE
D TO THE";:PRINT@837,"LOUDNESS S
ENSATION THIS WAVE";
240 PRINT@901,"MIGHT PRODUCE";
245 GOSUB1000
250 FORN=1TO128:RESET(N,A2(N)):N
EXT
255 PRINT@388,"AMPLITUDE";CHR$(9
4);
260 PRINT@645,"THE SECOND WAVE P
ATTERN";
265 PRINT@709,"HAS LESS AMPLITUD
E THAN THE";:PRINT@773,"1ST.  IT
 WILL PROBABLY";
270 PRINT@837,"BE LESS LOUD.";
275 GOSUB1000
280 CLS
285 PRINT"THEREFORE, THE AMOUNT
OF ENERGY IN A VIBRAATION CORRES
PONDS"
290 PRINT"TO OUR SENSATION OF LO
UDNESS."
295 PRINT"THE NUMBER OF TIMES A
VIBRATION REPEATS ITSELF EACH SE
COND IS ALSO IMPORTANT"
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```
300 PRINT"THIS IS KNOWN AS THE F
REQUENCY -- IT IS RELATED TO OUR
 SENSATION OF PITCH"
305 PRINT"THE MORE VIBRATIONS TH
ERE ARE EACCH SECOND, THE HIGHER
 THE PITCH"
310 PRINT:PRINT"IN OUR GRAPH OF
THE FORM OF VIBRATION JUST PRESE
NTED,"
315 PRINT"THE HEIGHT WAS THE REP
RESENTATION OF THE AMOUNT OF ENE
RGY.";
320 PRINT"WE CAM REPRESENT TIME
ACROSS THE LENGTH OF THE GRAPH:"

325 GOSUB1000
330 CLS
335 FORX=1TO120
340 SET(X,A(X))
345 NEXT
350 DATA A,M,P,L,I,T,U,D,E:FORX=
1TO9:READB$(X):NEXT
355 A=0:FORX=192TO704STEP64:A=A+
1:PRINT@X,B$(A);:NEXT
360 C$="TIME":D$=STRING$(4,94)
365 PRINT@960,C$;:FORN=1TO7:PRIN
TD$;C$;:NEXT
370 FORX=1TO120:SET(X,23):NEXT
375 PRINT@0,"A WAVEFORM WITH TWI
CE AS FAST A VIBRATION WOULD BE:
"
380 GOSUB1000
385 X=0:FORM=1TO2:FORM1=1TO60:X=
X+1:B=B+2:SET(X,A(B)):NEXTM1:B=0
:NEXTM
390 GOSUB2000
395 PRINT"THE SIMPLEST, SIME, VI
BRATION, CAM THUS HAVE TWO ASPEC
TS"
400 PRINT"IN OUR SENSTION:  PITC
H AND LOUDNESS":PRINT
405 PRINT"THIS SIMPLE VIBRATION
IS A BUILDING BLOCK FOR COMPLEX"

410 PRINT"VIBRATIONS -- ANY SOUN
D CAM BE BROKEN DOWN INTO A SERI
ES"
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415 PRINT"OF SIMPLE VIBRATIONS.
 ANY COMPLEX SOUND CAN BE CREATE
D"
420 PRINT"BY SOUNDING A SET OF T
HESE SINE WAVES SIMULTANEOUSLY"
425 PRINT:PRINT:PRINT"A THIRD SE
NSATION IS THE RREESULT OF THE A
MOUNT OF ENERGY"
430 PRINT"(AMPLITUDE) IN EACH SI
MPPLE VIBRATION COMPONENT OF A C
OMPLEX TONE:  TIMBRE."
435 PRINT:PRINT"TIMBRE IS THAT S
ENSATION THAT ALLOWS TWO TONES O
F EQUAL PITCH"
440 PRINT"AND LOUDNESS TO BE DIS
CRIMINATED FROM ONE ANOTHER"
445 GOSUB2000
450 PRINT"THE AMOUNT OF ENERGY I
N EACH SIMPLE COMPONENT OF A COM
PLEX"
455 PRINT"TONE PLOTTED AGAAINST
THE FREQUENCY OF THAT TONE IS CA
LLED"
460 PRINT"THE POWER SPECTRUM OF
THE TONE.  THE SENSATION OF TIMB
RE"
465 PRINT"IS PRIMARILY RELATED T
O THE POWER SPECTRUM"
470 GOSUB2000
475 PRINT"THIS INVESTIGATION WIL
L DETERMINE HOW SENSITIVE YOU AR
E"
480 PRINT"TO TIMBRE CHANGES.  TH
E TIMBRE CHANGES WILL BE INDUCED
"
485 PRINT"BY VARYING THE AMOUNT
OF ENERGY IN ONE SIMPLE COMPONEN
T"
490 PRINT"OF A COMPLEX SOUND.  Y
OU WILL COMPARE TWO-TONE PAIRS:
 ONE"
495 PRINT"WILL BE A STANDARD, TH
E OTHER WILL BE EITHER A STANDAR
D OR"
500 PRINT"COMPARISON, RANDOMLY P
RESENTED..  YOU SIMPLY DETERMINE
 IF THEY"
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505 PRINT"HAVE THE SAME OR DIFFE
RENT TIMBRE."
510 PRINT:PRINT"THE COMPARISON T
ONES WILL HAVE MORE AND MORE CHA
NGES"
515 PRINT"IN THE ENERGY IN EACH
SIMPLE TONE.  THE OVERALL ENERGY
"
520 PRINT"IN THE COMPLEX TONE RE
MAINS THE SAME, BECAUSE IF ONE C
OMPONENT"
525 PRINT"IS INCREASED, THE OTHE
R COMPONENTS ARE PROPORTIONALLY
DECREASED"
530 PRINT"IN ENERGY.":GOSUB2000
535 FORY=9TO32:SET(32,Y):NEXT:FO
RX=32TO93:SET(X,32):NEXT
540 PRINT@450," AMPLITUDE";
545 PRINT@722,"1   2   3   4
5   6   7";
550 PRINT@22,"STANDARD STIMULUS"
:H=27
555 FORG=1TO7:H=H+9:FORY=20TO32:
SET(H,Y):SET(H+1,Y):SET(H+2,Y):S
ET(H+3,Y)
560 NEXTY:NEXTG
565 IFR1<3ANDR1>0THEN610
570 IFR1=3THEN800
575 PRINT@768,"THE BARS IN THE G
RAPH REPRESENT THE AMPLITUDE"
580 PRINT"OF EACH SIMPLE COMPONE
NT OF THE STANDARD STIMULUS USED
"
585 PRINT"IN THIS INVESTIGATION
THEY ALL HAVE THE SAME AMPLITUDE
"
590 D1=1:GOSUB2000
595 PRINT@763,"THIS DEMONSTRATIO
N INDICATES HOW THE COMPARISON S
TIMULI"
600 PRINT@832,"WILL BE CREATED B
BY DISTRIBUTING THE ENERGIES AMO
NG"
605 PRINT@896,"THE OTHER PARTIAL
S"
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610 PRINT@22,"COMPARISON STIMULU
S":FORY=19TO17-R1STEP-1:SET(36,Y
):SET(37,Y):SET(38,Y):SET(39,Y)
615 NEXT
620 FORY=20TO21:FORX=44TO94:RESE
T(X,Y):NEXTX,Y
625 R1=R1+1
630 FORY=17-R1TO19:RESET(36,Y):R
ESET(37,Y):RESET(38,Y):RESET(39,
Y):NEXT
635 GOTO550
800 D1=0:GOSUB2000
805 PRINT"YOU NOW HAVE A GENERAL
 BACKGROUND IN THE PRINCIPLES"
810 PRINT"BEHIND TIMBRE AND COMP
LEX SOUNDS.  IF YOU HAVE ANY QUE
STIONS"
815 PRINT"REFER THEM TO THE EXP
ERIMENTER":GOSUB2000:PRINTCHR$(2
3):PRINT@462,"THANK YOU"
816 END
1000 PRINT@985,"PRESS ANY KEY TO
 CONTINUE";
1005 Z$=INKEY$:IFZ$=""THEN1005
1010 PRINT@645,"
            ";
1015 PRINT@709,"
             ";
1020 PRINT@773,"
            ";
1025 PRINT@837,"
            ";
1030 PRINT@901,"
            ";
1035 RETURN
1500 IP=IP+.5:FORN=1TOIP:NEXT:RE
TURN
2000 PRINT@985,"PRESS ANY KEY TO
 CONTINUE";
2001 Z$=INKEY$:IFZ$=""THEN2001
2002 IFB1=1THENRETURNELSECLS
2003 RETURN
3000 X1=-3:FORX=1TO128:X1=X1+3:Y
1=SIN(X1*.0174533):Y=24-(Y1*18):
A(X)=Y:NEXTX:RETURN
4000 X1=-3:FORX=1TO128:X1=X1+3:Y
1=SIN(X1*.0174533):Y=24-(Y1*13):
A2(X)=Y:NEXTX:RETURN
```