THE COMPUTER MUSIC and DIGITAL AUDIO SERIES

John Strawn, Series Editor

DIGITAL AUDIO SIGNAL PROCESSING: An Anthology
Edited by John Strawn
with contributions by J. W. Gordon, F. R. Moore, J. A. Moorer, T. L. Petersen, J. O. Smith, and J. Strawn

COMPOSERS AND THE COMPUTER
Edited by Curtis Roads
with contributions by or about H. Brün, J. Chowning, J. Dashow, C. Dodge, P. Hamlin, G. Lewis, T. Machover, J.-C. Risset, C. Roads, and I. Xenakis

DIGITAL AUDIO ENGINEERING: An Anthology
Edited by John Strawn
with contributions by J. McGill, F. R. Moore, J. A. Moorer, P. Samson, and R. Talambiras

THE LITTLE BOOK OF COMPUTER MUSIC INSTRUMENTS
Dexter Morrill

A COMPUTER MUSIC HISTORY
Curtis Roads

COMPUTER APPLICATIONS IN MUSIC: A Bibliography
Deta S. Davis

COMPOSERS AND THE COMPUTER

CURTIS ROADS, Editor

WILLIAM KAUFMANN, INC.
Los Altos, California

Herbert Brün. *Photograph by David Bunn.*
John Chowning. *Photograph by Patte Wood.*
James Dashow. *Photograph by Gordon Hallberg.*
Charles Dodge. *Photograph by Gene Bagnato.*
George Lewis. *Photograph by Jean-Pierre Plisson.*
Tod Machover. *Photograph by Anne Marie Stein.*
Jean-Claude Risset. *Photograph by Rozenn Risset.*
Curtis Roads. *Photograph by Michael Bate.*
Iannis Xenakis. *Photograph by J. V. III.*
Cover image. *Henkjan Honing.*
Music printing. *Leland Smith.*

Library of Congress Cataloging in Publication Data
Main entry under title:

Composers and the computer.

(The Computer music and digital audio series)
Bibliography: p.
Includes index.
1. Computer music—History and criticism. 2. Computer composition.
I. Roads, Curtis. II. Series.
ML1092.C65 1985 789.9'9 84-23323
ISBN 0-86576-085-3

William Kaufmann, Inc.
95 First Street
Los Altos, California 94022

Printed in the United States of America

10 9 8 7 6 5 4 3 2 1

CONTENTS

THE AUTHORS

Herbert Brün is on the faculty of music at the University of Illinois at Urbana-Champaign.

John Chowning is on the faculty of music at Stanford University, Stanford, California, and is director of the Center for Computer Research in Music and Acoustics there.

James Dashow is director of Studio Sciadoni, Rome, and is currently finishing an opera entitled *Il Piccolo Principe.*

Charles Dodge is on the faculty of music at Brooklyn College, City University of New York.

Peter Hamlin is a broadcaster with KPBS-FM, San Diego.

George Lewis is a researcher at the Institut de Recherche et Coordination Acoustique/Musique and a musician/composer in Paris.

Tod Machover is on the faculty of the Arts and Media Technology facility at the Massachusetts Institute of Technology.

Curtis Roads is a research associate at the Massachusetts Institute of Technology and editor of *Computer Music Journal,* published by MIT Press.

Jean-Claude Risset is on the faculty of physics and acoustics at the University of Marseille.

Iannis Xenakis is director of the Centre des Etudes Mathématique et Automatique Musicales, Paris.

A NOTE ABOUT THIS SERIES

The Computer Music and Digital Audio Series has been established to serve as a central source for books dealing with computer music, digital audio, and related subjects.

During the past few decades, computer music and digital audio have developed as closely related fields that draw from a wide variety of disciplines: computer science, electrical engineering (especially digital signal processing and hardware), psychology (especially perception), physics, and all aspects of music.

The series includes, but is not limited to

- textbooks (at the undergraduate and graduate levels)
- how-to books (such as collections of patches for synthesis)
- anthologies (such as the current volume)
- reference works and monographs
- guides for audio engineers and studio musicians
- books for home computer users and synthesizer players

The present volume is one of three anthologies published in 1985 to inaugurate the series. Together, they cover subject matter that has not been covered in such breadth and depth. A number of other manuscripts are already in preparation; we anticipate that two or three volumes will appear each year for the next several years.

In an anthology such as this, it seemed important to let each composer speak for himself. These articles reflect each composer's views and feelings, just as a composition would. The unity of the whole anthology may thereby be weakened slightly, but the variety results in a richer statement of the current state of the art.

John Strawn
Stanford, California

INTRODUCTION

This book is devoted to composers and the issues they face in working with computers. It complements the material presented in The Computer Music and Digital Audio Series and in two other books coedited with John Strawn, *Computer Music Tutorial* and *Foundations of Computer Music* (published by MIT Press). Because these books are primarily concerned with technical aspects of computer music, *Composers and the Computer* was prepared to help satisfy the growing need for information about aesthetic issues and compositional techniques

For practical reasons, this book contains texts from only a few of the many composers working in the field. Although the views expressed here cover a broad spectrum, in no way does this collection purport to represent all significant composers or texts on computer music composition. Other important statements on composition can be found by culling the pages of publications such as *Computer Music Journal.*

In the rest of this introduction, I trace some major themes within this volume and try to convey a sense of the compositional environment out of which these composers have emerged. (See also Battier [1981] and Schwartz [1973].) The names of the composers appearing in this book are marked in italics at first mention.

The Historical Context

Current trends in composition have been prepared by broader historical processes. The fantastic "sound-houses" conjured up by Francis Bacon in *The New Atlantis* (1627) are one indication that the musical possibilities made possible with digital techniques have been imagined for centuries. (See Roads [forthcoming] for an account of computer music's historical background.) Access to an expanded universe of sound was a persistent dream of early twentieth-century composers (Busoni 1911). The quest for new electronic instruments was well articulated by the composer Edgard Varèse in 1939:

> Here are the advantages I anticipate from such a machine: liberation from the arbitrary, paralyzing tempered system; the possibility of obtaining any number of cycles or, if still desired, subdivisions of the octave, consequently the formation of any desired scale; unsuspected range in low and high registers; new harmonic splendors obtainable from the use of subharmonic combinations now impossible; the possibility of obtaining any differentiation of timbre, of sound combinations; new dynamics far beyond the present human-powered orchestra; a sense of sound-projection in space by means of the emission of sound in any part or in as many parts of the hall as may be required by the score; cross-rhythms unrelated to each other, treated simultaneously. . .all of these in a given unit of measure or time which is humanly impossible to obtain. (Varèse 1966)

After nearly half a century, with the availability of digital techniques for the analysis, synthesis, and processing of sound, this dream has materialized. Beyond the synthetic splendors imagined by Varèse, the possibilities offered by digital recording and processing techniques are enormous. Many composers are attracted to computer music systems in which they can digitize, process, and edit natural sounds such as those from traditional instruments (Banger and Pennycook 1983; Roads 1983). This activity follows the experiences of *tape music* (Ussachevsky 1960) and *musique concrète* (Schaeffer 1966), but goes beyond classic tape manipulations. Digital techniques offer the following advantages:

- Precise and rehearsable splicing and crossfading
- Noise-free mixing, looping, and dubbing
- Independent pitch and duration changing
- Replication of one sound into a chorus
- Highly selective echo and reverberation effects
- Precise control of spatial location
- Cross-synthesis, i.e., using the characteristics of one sound to shape the spectrum of another
- Continuous timbral interpolation from one sound to another

Such capabilities clearly point to new compositional directions (Harvey 1981; Haynes 1982; Roads 1983).

But digital studio techniques represent only part of the story. The use of computers, computer-controlled synthesizers, and digital hardware in live performance—pioneered in the early 1970s by Peter Zinovieff, Edward Kobrin, Salvatore Martirano, and Donald Buchla—has greatly increased in the 1980s. Digital processing in the performance of recent instrumental works by Morton Subotnick, Pierre Boulez, and Luciano Berio builds on earlier analog practices, but goes beyond these techniques to offer numeric and symbolic manipulations possible only in the digital domain. The ability of computers to listen and respond to *music,* and not just to *sound,* represents a qualitative change from previous analog electronic music possibilities.

Not all that is offered by computers is new. In some cases, the new possibilities are extensions or even rediscoveries of existing knowledge. For example, Robert Erickson's book (1975) contains a wealth of knowledge on sound manipulation techniques applied in past instrumental works. Many of these techniques have yet to be adapted to the computer music domain. Similarly, recent computer-based research on scale systems rests on a music-theoretical foundation laid in antiquity. Scientists such as Mathews and Pierce (1980) have retraced the steps of Pythagoras, Rameau, and Helmholtz in their studies of temperaments and scales. Composers such as Clarence Barlow (1980), who use the computer to synthesize microtonal compositions, follow a path opened up earlier in the twentieth century by Haba, Carillo, Vyschnegradsky, Ives, and Partch, among others. Similarly, some of the complex rhythmic exercises composed for computer in recent years owe a debt to past work by Joseph Schillinger, Henry Cowell, Olivier Messiaen, Conlon Nancarrow, and Elliot Carter, among many others.

Some digital techniques are little more than precise versions of previous mechanical or analog processes. But precision is not the most important attribute of the computer. What the computer offers the composer is programmability—the extension of functionality in any direction. The computer can be used to control a synthesizer, to process sounds, to edit scores, to create scores according to composer-specified rules, to print music, to analyze music, and to act as a partner in improvisation.

The Computer as Musical Instrument

To some composers, notably *Charles Dodge* and *Jean-Claude Risset,* the computer is, above all, a wonderfully pliable instrument. It offers sound combinations and manipulations unobtainable by other means. Dodge and Risset are musical craftsmen, among the first to prove the artistic viability of the medium to the musical world.

Taking Arnold Schoenberg's *Sprechstimme* one step further, Dodge's specialty is the setting of text, spoken and sung by computer, as in his humorous *Speech Songs* (1973) and *Any Resemblance Is Purely Coincidental* (1981). His article here discusses both the compositional organization of the more somber *In Celebration* (1976) and the linear predictive coding (LPC) synthesis method used to realize it. The LPC technique, originally developed for speech synthesis, has been used in a series of effective compositions by Dodge and others such as Paul Lansky (Roads 1983).

Another trend in composition is the blending of sounds produced by traditional instruments, the human voice, and digital computers or synthesizers (Morrill 1981). Jean-Claude Risset is a master at combining traditional instruments and idioms (such as twelve-tone organization) with purely digital sounds (for example, sine-wave clusters and glissandi), and composition techniques facilitated by computers (for instance, manipulation of timbre space). *Passages* for flute and tape, premiered at the 1982 International Computer Music Conference in Venice (reviewed in Blum et al. [1983]), is a prime example of virtuosic blending of computer sound with traditional instruments. Risset's longtime interest in the microstructure of sound is reflected in many of the proposals made in his paper.

Tod Machover, in his contribution, also expresses a keen interest in the possibilities of synthesis and processing of natural and artificial sound. He explores this interest in a variety of ways in his compositions *Déplacements* (1979), *Light* (1979), and *Soft Morning, City!* (1980). One of the contrasts brought out in these discussions is the emphasis on real-time and interactive work espoused by Machover versus the stress on non-real-time experimentation championed by Risset. Machover also believes that automated composition techniques may be useful, particularly in an interactive synthesis environment.

Procedural Composition

In addition to a dream of an extended palette of sounds, many musicians and musical scholars throughout history have been fascinated with the idea of *musical process* and *musical procedures,* whether in theory or in mechanical form. The composition of music according to procedures has a long history (Kircher 1650; Hiller 1970). Recent computer-based experiments were antedated by Guido d'Arezzo's table lookup procedure for assigning vowels to pitches (c. 1030), by Affligemensis's rules (c. 1130) along the same lines, and by the musical games of S. Pepys (1670) and W. A. Mozart (1770). Another important development was D. Winkel's *Componium* (completed in 1821)—a mechanical contraption for producing variations on themes programmed into it.

Computer technology is an ideal medium for implementing a procedural or *algorithmic* approach to composition. (An algorithm is an explicit, finite procedure for accomplishing a task.) It allows the composer to encode a musical idea as software—a computer program (Ames 1982, 1983; Barbaud 1966; Hiller 1981; Hiller and Isaacson 1959; Smoliar 1971; Xenakis 1971). With digital technology composers can combine specifications for rich and intricate sound formations with abstract specifications for musical process (Berg 1979; Chadabe and Meyers 1977; Englert 1981; Holtzman 1981; Jones 1981; Kendall 1981; McNabb 1981).

G. M. Koenig, for example, poured years of research into his Project 1 and Project 2 composing programs (Koenig 1970a, 1970b; Laske 1983; Roads 1978). These programs can be seen as the codification by a master practitioner of the post-Webern sensibility that flowered in Europe in the 1950s and 1960s.

Herbert Brün pursues a unique approach to computer music. For Brün, composition is not just a musical act, it is also an expression of philosophy, as the interview by Peter Hamlin shows (see also Brün [1969, 1973]). With Brün's SAWDUST system, the composer works interactively with the computer to construct entities that can be either individual sound objects or collections of objects. In both SAWDUST and Koenig's SSP (Berg 1978), the basic elements are time and amplitude points, out of which are constructed simple waveforms and, ultimately, large-scale musical structures.

In keeping with the tradition of experimental music, both Koenig and Brün have at times cultivated a detached attitude toward the audible result of their composing labors. They concentrate on the generative system; the sound produced by it is a by-product. For this reason, their music calls for a different kind of listening, based on attention to musical algorithms and processes.

Interaction with Programming Languages

In *John Chowning*'s approach to composition, the programming language is more than just a passive medium for articulating ideas. Rather, it shapes and can even inspire musical expression. In the interview in this book, he discusses his approach to composition and synthesis in three works: *Turenas* (1971), *Stria* (1978), and *Phōnē* (1981). *Turenas* and *Phōnē* are based on frequency modulation (FM) synthesis, while *Stria* uses additive combinations of pure sine waves. Chowning also recounts his pioneering research on the movement of sounds in space, FM synthesis, and voice synthesis (Chowning 1971, 1973, 1980).

James Dashow is a virtuosic manipulator of synthesis languages, which he adapts to realize his unique compositional vision. As the interview here reveals, Dashow uses synthesis languages to embed musical ideas into the process of sound generation itself. For Dashow, the computer provides an opportunity to merge pitch and interval manipulations with timbral processing. (See also Dashow [1980] for a purely technical description.) Contrary to the practices of some composers, Dashow makes no attempt to simulate traditional instruments. Compositions such as *In Winter Shine* (1983) depend on purely synthetic timbres and effects possible only in the digital domain.

Indeterminacy and Improvisation

One topic broached in a number of the articles in this book is that of *systematic* versus *improvised* approaches to composition—an ancient dichotomy that continues to be contested. One of the most strenuous debates over this issue took place in the 1950s and 1960s when several of the composers featured here were forming their compositional outlook. John Cage's influential book *Silence* (1961) promulgated the notion of *chance* elements in composition, and the whole question of how to handle indeterminacy held the attention of many prominent composers such as Pierre Boulez, Karlheinz Stockhausen, and *Iannis Xenakis*.

Boulez's solution to this problem (e.g., *Trope* [1961]) was to introduce *mobile form* (also pioneered by the American composer Earle Brown), in which the performer selects the ordering of already-composed material (Boulez 1971). Stockhausen ultimately decided to distinguish between *improvisation* (in which a schema is filled in by performers) and *intuitive music* (e.g., *Aus den Sieben Tagen* [1969]), in which the performers work from no formal structure except that suggested by a poetic text (Stockhausen 1971).

One of the first contributions of Iannis Xenakis addressed the question of indeterminacy. This was the concept of *stochastic* music, heard in pieces such as Xenakis's *ST/10-1, 080262* for ten instruments. Lejaren Hiller (Hiller and Isaacson 1959; Hiller 1981) and Pierre Barbaud (Barbaud 1966) also pioneered the use of probabilistic techniques. In the stochastic compositions of the 1950s through the 1970s, chance was allowed to play at the microstructural level (i.e., at the level of notes, note parameters, or waveforms), whereas larger-scale musical structures remained under the composer's direct control.

But many paths to composition have been traced by Xenakis. His essay "Music Composition Treks" in this collection deals at once with broad philosophical issues as well as specific compositional ideas. Like John Chowning, Xenakis feels composers must acquire requisite technical knowledge, including computer literacy and a fundamental knowledge of mathematical

concepts, to fully exploit the computer medium. Unlike Chowning, however, Xenakis has no faith in the musical application of traditional signal-processing operations such as Fourier analysis.

Another area explored by Xenakis is the integration of visual and musical media through computer technology. As anyone who has experienced one of his *Polytopes* or *DIATOPEs* knows, the coordination of architectural space with sound and laser light can create a dazzling spectacle.

Intelligent Instruments in Performance

Digital techniques make possible intelligent instruments for live performance. These new instruments allow musicians to combine preprogrammed and spontaneous gestures in concert (Abbott 1981). Certain of these instruments can "listen" to music, "watch" a conductor, and respond to sensed musical contexts.

Once again, the compositional problem of coordinating preplanned versus improvised musical structure comes to the fore. In facing this problem, composers sometimes draw from solutions provided by world music traditions. For example, the composer/trombonist *George Lewis* treats the computer as a partner in improvisation. Listening, as well as playing, is important in improvisation, and Lewis's improvising program for a small computer embodies both a listening and a sound-generating component. The computer listens via a microphone and performs by means of a digital synthesizer connected to the computer.

Another composer, Joel Chadabe, bases his concept of "interactive composing" on the interaction of the composer with unique input devices such as Theremin antennae and special drum pads (Chadabe 1984). Software translates these gestures into high-level musical processes. With Chadabe's Solo system, a wave of the composer's hand can cause the tempo of a musical process to change and the harmony to shift.

Interactive microcomputer music experimentation flourished in the 1970s at places such as Mills College in California and the University of Paris VIII—Vincennes (through artists affiliated with the Groupe Art et Informatique Vincennes). Since then, the view of computer music as a rare and costly musical technology has faded. Many digital instruments can be attached to inexpensive personal computers. The proliferation of inexpensive computers puts the capability of intelligent instruments within the reach of virtually every musician who wants them.

Multimedia Performance

Speech-oriented computer music often blurs the boundary between poetry, theater, and music. The interaction of computer music with other media, such as film, video, slide projections, lasers, dance, poetry, theater, performance art, and even architecture, is another domain that has already yielded important artistic results. Examples include Xenakis's *DIATOPE* and Dashow's opera *Il Piccolo Principe;* the latter utilizes seven singers, four channels of digital sound, computer graphics, and laser projections. The use of the computer in interactive music-theater performances is also documented (Cavaliere et al. 1982). The possibility of creating in several media using an integrated computer system is especially promising. The advantages of a system that could combine music and image-making, or music and choreography, are obvious.

Theory and Current Practice

Computers have already affected our sense of music. The expanded sound palette made available by new technology has prompted a reassessment of musical structure and terminology. The trinity of "melody, harmony, and rhythm" and other concepts such as scales, tonalities, and timbres are, as ever, available to the computer music composer. However, as my paper on *nscor* discusses, a wealth of sound exists for which traditional musical concepts such as the note (to give one example) seem inadequate. The note has given way in some cases to the notion of *sound objects* (Schaeffer 1966) or active *processes* (Schottstaedt 1983; Rodet and Cointe 1984). Here theory lags far behind compositional practice.

Conclusion

At the present time, composers differ as to the degree the computer influences their musical thinking. In any case, a fundamental lesson of the papers assembled here is that available technology can strongly affect the working practice of composition. This is especially the case for the composers represented here, who take a "hands-on" approach to the technology, without the need for assistants or other intermediaries. Imminent technical advances in the areas of musician-machine interface, real-time hardware, signal processing, and artificial intelligence are sure to open up new possibilities and thus have a further impact on the way we compose.

Acknowledgments

This book began as a group of essays originally intended as a chapter in a larger collection. William Kaufmann and John Strawn created the opportunity to turn this text into a book on its own. In addition, John Strawn read over this manuscript with characteristic thoroughness and offered many valuable suggestions.

All royalties for this book have been assigned to Oxfam America, Inc., a nonprofit famine relief organization.

Curtis Roads
Cambridge, Massachusetts

References

Abbott, C. 1981. The 4CED program. *Computer Music Journal* 5(1):13–33.

Ames, C. 1982. Crystals: Recursive structures in computer music composition. *Computer Music Journal* 6(3):46–64.

Ames, C. 1983. Stylistic automata in *Gradient. Computer Music Journal* 7(4):45–56.

Bacon, F. 1627. *The New Atlantis.* Reprinted in R. Coffin and A. Witherspoon, eds. 1929. *A book of seventeenth century prose,* 37–55. New York: Harcourt, Brace, and Co.

Banger, C., and B. Pennycook. 1983. GCOMP: Graphic control of mixing and processing. *Computer Music Journal* 7(4):33–39.

Barbaud, P. 1966. *Initiation à la composition algorithmique.* Paris: Dunod.

Barlow, C. 1980. Bus journey to parametron. *Feedback Papers* 21/23:1–124.

Battier, M. 1981. Les tendences récentes des musiques électroacoustiques et l'environment informatique. Ph.D. diss., Univ. of Paris VIII.

Berg, P. 1978. A user's manual for SSP. Utrecht: Institute of Sonology.

Berg, P. 1979. PILE: A language for sound synthesis. *Computer Music Journal* 3(3):30–41. Reprinted in C. Roads and J. Strawn, eds. 1985. *Foundations of computer music.* Cambridge: MIT Press.

Blum, T., et al. 1983. Report on the 1982 International Computer Music Conference. *Computer Music Journal* 7(2):8–35.

Boulez, P. 1971. *Boulez on music today.* S. Bradshaw and R. Bennett, trans. Cambridge: Harvard Univ. Press.

Brün, H. 1969. Infraudibles. In H. Von Foerster and J. Beauchamp, eds. 1969. *Music by computers,* 117–21. New York: Wiley.

Brün, H. 1973. *Über Musik und zum Computer.* Karlsruhe: G. Braun Verlag.

Busoni, F. 1911. *Sketch of a new esthetic of music.* In T. Baker, trans. 1962. *Three classics in the aesthetics of music,* 73–102. New York: Dover.

Cage, J. 1961. *Silence.* Middletown, Conn.: Wesleyan Univ. Press.

Cavaliere, S., et al. 1982. From computer music to theater: The realization of a theatrical automaton. *Computer Music Journal* 6(4):22–35.

Chadabe, J. 1984. Interactive composing. *Computer Music Journal* 8(1):22–27.

Chadabe, J., and R. Meyers. 1977. An introduction to the PLAY program. *Computer Music Journal* 1(4):12–18. Reprinted in C. Roads and J. Strawn, eds. 1985. *Foundations of computer music.* Cambridge: MIT Press.

Chowning, J. 1971. The simulation of moving sound sources. *Journal of the Audio Engineering Society* 19(1):2–6. Reprinted in *Computer Music Journal* 1(3):48–52, 1977.

Chowning, J. 1973. The synthesis of complex audio spectra by means of frequency modulation. *Journal of the Audio Engineering Society* 21(7):526–34. Reprinted in C. Roads and J. Strawn, eds. 1985. *Foundations of computer music.* Cambridge: MIT Press.

Chowning, J. 1980. Synthesis of the singing voice by frequency modulation. In E. Jansson and J. Sundberg, eds. 1980. *Sound generation in winds, strings, and computers,* 4–13. Pub. No. 29. Stockholm: Royal Swedish Academy of Music.

Dashow, J. 1980. Spectra as chords. *Computer Music Journal* 4(1):43–52.

Englert, G. 1981. Automated composition and composed automation. *Computer Music Journal* 5(4):30–35.

Erickson, R. 1975. *Sound structure in music.* Berkeley: Univ. of California Press.

Harvey, J. 1981. *Mortuos Plango, Vivos Voco:* A realization at IRCAM. *Computer Music Journal* 5(4):22–24.

Haynes, S. 1982. The computer as a sound processor: A tutorial. *Computer Music Journal* 6(1):7–17.

Hiller, L. 1970. Music composed with computers—A historical survey. In H. Lincoln, ed. 1970. *The computer and music,* 42–96. Ithaca, N.Y.: Cornell Univ. Press.

Hiller, L. 1981. Composing with computers: A progress report. *Computer Music Journal* 5(4):7–21.

Hiller, L., and L. Isaacson. 1959. *Experimental music.* New York: McGraw-Hill.

Hiller, L., and R. Baker. 1964. *Computer cantata:* A study in compositional method. *Perspectives of New Music* 3(1):62–90.

Holtzman, S. 1981. Using generative grammars for music composition. *Computer Music Journal* 5(1):51–64.

Howe, H. S., Jr. 1975. *Electronic music synthesis.* New York: Norton.

Jones, K. 1981. Compositional applications of stochastic processes. *Computer Music Journal* 5(2):45–61.

Kendall, G. 1981. Composition from a geometric model: *Five-leaf rose. Computer Music Journal* 5(4):66–73.

Kircher, A. 1650. *Musurgia universalis.*

Koenig, G. M. 1970a. Project One. *Electronic Music Reports* 2. Utrecht: Institute of Sonology. Reprinted 1977. Amsterdam: Swets and Zeitlinger.

Koenig, G. M. 1970b. Project Two. *Electronic Music Reports* 2. Utrecht: Institute of Sonology. Reprinted 1977. Amsterdam: Swets and Zeitlinger.

Laske, O. 1981. Composition theory in Koenig's Project 1 and Project 2. *Computer Music Journal* 5(4):54–65.

McNabb, M. 1981. *Dreamsong:* The composition. *Computer Music Journal* 5(4):36–53.

Mathews, M., and J. R. Pierce. 1980. Harmony and nonharmonic partials. *Journal of the Acoustical Society of America* 68:1252–57.

Morrill, D. 1981. Loudspeakers and performers: Some problems and proposals. *Computer Music Journal* 5(4):25–30.

Roads, C. 1978. An interview with Gottfried Michael Koenig. *Computer Music Journal* 2(3):11–15.

Roads, C. 1981. A report on the IRCAM Conference: The composer and the computer. *Computer Music Journal* 5(3):7–27.

Roads, C. 1982. Music and artificial intelligence: A research overview. *Proceedings of the Italian Computer Society,* 401–6. Padua: Associazione Italiana per il Calcolo Automatico.

Roads, C. 1983. Interview with Paul Lansky. *Computer Music Journal* 7(3):16–24.

Roads, C. Forthcoming. *A computer music history.* Los Altos, Calif.: Kaufmann.

Rodet, X., and P. Cointe. 1984. FORMES: Composition and scheduling of processes. *Computer Music Journal* 8(3):32–50.

Schaeffer, P. 1966. *Traité des objets musicaux.* Paris: Seuil. Rev. ed. 1977.

Schottstaedt, B. 1983. Pla: A composer's idea of a language. *Computer Music Journal* 7(1):11–20.

Schwartz, E. 1973. *Electronic music: A listener's guide.* New York: Praeger.

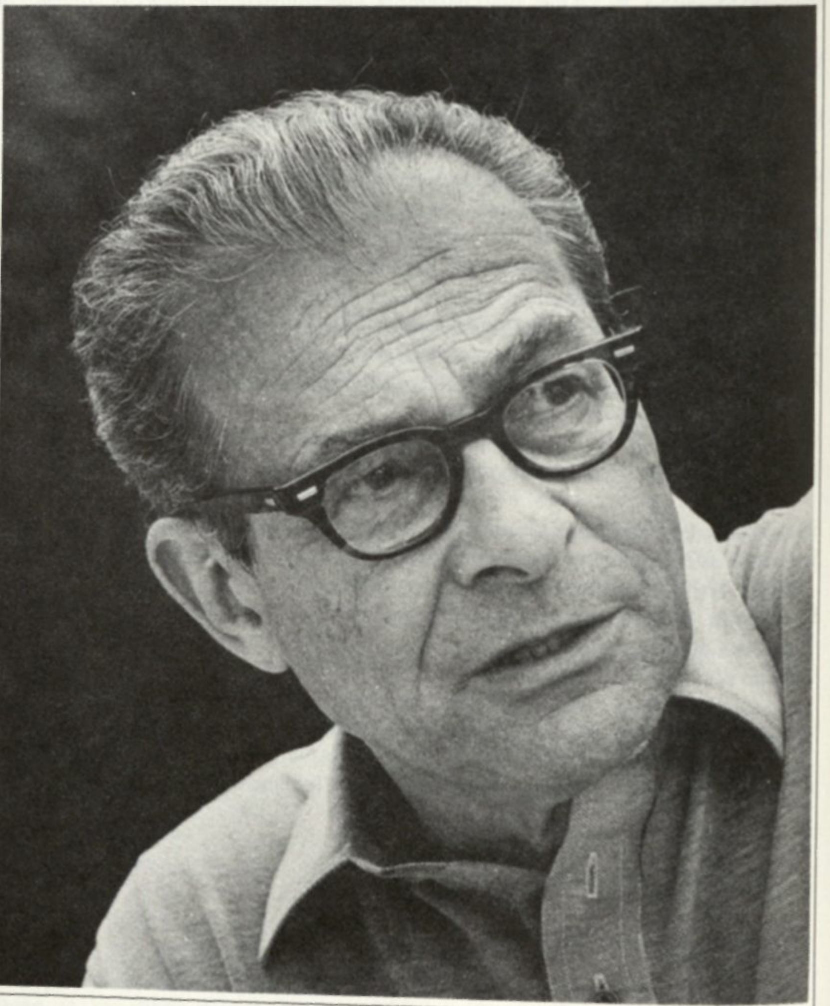
Smoliar, S. 1971. A parallel processing model of musical structures. AI-TR-242. Cambridge: Artificial Intelligence Laboratory, MIT.

Stockhausen, K. 1971. Aus den Sieben Tagen. In *Texte zur Musik: 1963–1970.* Vol. 3, 123–34. Cologne: DuMont.

Ussachevsky, V. 1960. Notes on a piece for tape recorder. In P. H. Lang, ed. *Problems of modern music,* 64–71. New York; Norton.

Varèse, E. 1966. The liberation of sound. In B. Boretz and E. Cone, eds. 1971. *Perspectives on American composers,* 25–33. New York: Norton.

Xenakis, I. 1971. *Formalized music.* Bloomington: Indiana Univ. Press.



HERBERT BRÜN

1

INTERVIEW WITH HERBERT BRÜN

Peter Hamlin
with Curtis Roads

Herbert Brün's efforts in computer music composition have centered, in recent years, on a system called SAWDUST. Work on the system began in 1973 and has continued ever since. SAWDUST was designed for waveform synthesis, making it possible to compose, transform, and play waveforms in an interactive manner.

The commands provided with the SAWDUST system were described in detail by T. Blum (1979). As of 1984, the list of commands includes the following:

```
ELEMENT
LINK
MINGLE
MERGE
VARY
TURN
PLAY
SHOW
```

Brün's work with the SAWDUST system has resulted in seven compositions so far:

Dust	1976	(9:30)
More Dust	1977	(12:45)
More Dust with Percussion	1979	(13:45)
Dustiny	1978	(5:45)
A Mere Ripple	1979	(11:36)
U-Turn-To	1980	(6:00)
I Told You So	1982	(13:00)

Dust and *More Dust* are available on a tape entitled *Herbert Brün: Project Sawdust* from the Lingua Press, P.O. Box 481, Ramona, CA 92065. A three-record set of the compositions of Herbert Brün is available from Non Sequitur Records, Box 872, Champaign, IL 61820.

The following interview took place during the afternoon of 30 October 1977, on the campus of the University of California, San Diego, in La Jolla. Professor Brün lectured the day before on his SAWDUST system, and played one of its first results, a composition entitled *Dust* (1977).

Hamlin: Is it Doctor or Mister?

Brün: Mister. My first appearance in the academic world was 1963 at the University of Illinois, invited by Lejaren Hiller, who was looking for someone to join him in his research with the computer and music. I had such a project in Munich, but he had the computers and I didn't have the computers. I wrote a little paper; he read it and called me to "come and see" for a year. I did "come and see" for a year and stayed there. Now I'm a professor on the faculty.

Hamlin: Some of the people who have gotten into this field are musicians who became interested in computers; others are technologists who became interested in music. Which direction are you coming from?

Brün: I come from the direction of having had some thoughts about the concept of composition. It started after the Darmstadt experiences and experiences at the electronic music studio in Cologne. Certain things began to become redundant. Even though they were fascinating and intriguing, they began to behave like a mathematical group—that is, no matter how I permuted them or otherwise operated on them, they always recreated some member of the same group. I had to catapult myself out of that loop.

So I was looking for some interaction, be it with people or with a medium, or even with myself in a hitherto untried way, where I could find a kind of slingshot situation: where the loop becomes so fast that when I let go I am thrown out of it.

I started writing a score for orchestra in which I used the method of having tables and precompositional material ready on the walls and on the table and on the floor—to an absurd state of completeness. I got, as could be predicted, totally stuck—confused. It was not really an unhappy affair, but it was a puzzling situation. At that point, I decided I had to make an experiment: I had to find out whether I could compose a structure without a look at the system to which I could apply it. So what I needed was a noncommitted system, which would allow me to first program a structure and then to fill the empty system with some stipulated members of various kinds—always applying the same structure to different sets of stipulated members. I wanted first of all just to see what would happen. That was the beginning; it comes from composition. Let's put it this way: from a middle-class-bourgeois-linguistic environment in which the words *composition* and *music* have a strong relationship, I came to a point where I discriminated between the two radically. I said: music is *traces left by composition* and not identical with composition.

Hamlin: I want to back up a bit to see if I've got you right. You were trying to hurl yourself from this loop by removing yourself to some degree from the sounds you were creating . . .

Brün: That's correct. I knew by that time that I was a talented musician, also that my heritage—the philharmonic concerts, the record collections, the education at home, in school, and piano lessons—had provided me with a lexicographical knowledge of tunes, harmonic progressions, and timbres. So I always wallowed in a world which took over whenever *I* wanted to do something. It offered itself to me ingratiatingly, again and again and again. I got tired of that.

So I had to find some way to affectionately liberate myself from myself, nevertheless still distinguishing myself from and in a society which I don't find yet desirable.

Hamlin: Are the 1970s a particularly bad time for having things within you that you don't really want within you? I'm not talking only about a mass of music history that we all hear but also a mass of commercial music. Is this a particularly bad time for all this kind of thing?

Brün: If I accept your vocabulary I would say, yes, it is a particularly bad time. If I don't accept your vocabulary I would say, please leave out the word "bad" and ask: "Is it a time for such things?" And then I would say, yes, it is a time for such things, a time when we allow ourselves to be entertained by obsolete treasures and communicative junk, call that "culture" and are mocked while we applaud it. If you would call "bad" a moment of utter need, as we also say when five thousand or five million people are starving, then we call that "bad times." We call them "bad" times because there is an urgent need not being satisfied, and we call them bad "times" because we like to pretend that they happen to us instead of admitting that we happen to them. As if it were not we who are needed, but merely better times.

Thus, to be an input to our times, our cultures, I am doing what is needed. I'm *not* the only one who needs it. I think *it is needed,* and so, as long as I can do something about it, I am needed. In this, I am in conformity with almost every human being. One of the basic needs of a human being is to be needed. So I found a way to feel needed, to tell myself occasionally, once a week or twice, no matter how freakish it is what I am trying to do, it is based upon some awareness of my being needed very much.

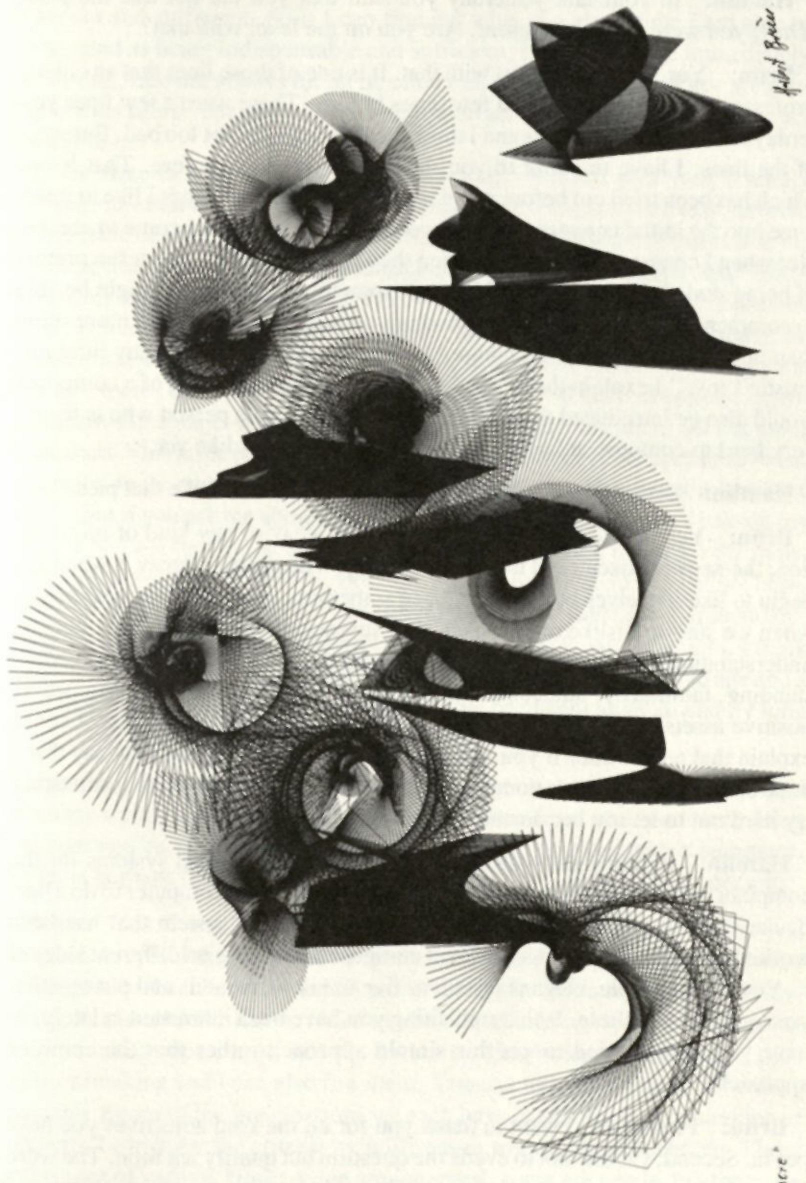


Fig. 1.1. "more": computer graphic by Herbert Brün

Hamlin: In your talk yesterday you said that you did not like the piece [*Dust*] you were going to present. Are you on the level with that?

Brün: Yes, I'm on the level with that. It is one of those lines that an elderly professor will have rehearsed a few times before. There were a few lines yesterday that were spontaneous and I think they were really not too bad. But some of the lines, I have to admit to you, have been tried out before. That is one which has been tried out before and it is one of the first sentences I like to introduce into the initial conversation with young people when they come to see me. Not when I come to see them, but when they come to see me, under the pretext of being students, or wanting to show me something, believing I might be able to comment in a valuable way. We embark on a conversation, and, more often than not, we come to a point where I have an opportunity to use my little linguistic "toy." I explain that it would be nice if in the definition of a composer would also be introduced the notion that a composer be a person who is trying very hard to compose at last the music he or she doesn't like yet.

Hamlin: Do you think a time will come when you will like that piece?

Brün: Yes, it will. There are two kinds of decay of any kind of information, the negative decay and the positive decay. The positive decay is when we begin to like ourselves in the presence of something, and the negative decay is when we start to dislike ourselves in the presence of something, due either to understanding or familiarity, or communicativity. These three things: understanding, familiarity, and communicativity, are not to be taken for granted as positive assets of our social interactions. They are dangerous things. I could explain that a little later if you ask me. At the moment I would only say that I hope always to compose a composition which teaches me the next aesthetics. I try hard not to let my last aesthetics compose my next piece.

Hamlin: In doing that you have developed very personal systems for the computer, very personal ways of communicating with the computer to do a particular task, as opposed to taking a massive computer system that has been worked on for ten years with different composers working on different sides of it. You take a simple, elegant system of five or six commands and put together your sounds with these. Is this something you have been interested in lately, or have you always tried to get this simple approach rather that the complex approach?

Brün: First of all, I want to thank you for all the kind adjectives you have put in. Second, I'd like not to evade the question but qualify it a little. The word "always" is certainly uncalled for. I don't do anything always, as far as I know. There are certainly things I do always and I hope I don't know them; they are very unpleasant. But I agree with you; I could concede that I have a strong *preference* for the indispensable and sufficient in contradistinction to the abundant and sufficient. So if I can find out what at a given time I am able to understand as being indispensable and sufficient, I will of course immediately prefer that. I do not always hit on it; often I also commit errors. They are usually not disasters. The attitude is more important to me than whether I'm successful in it.

Then there is one other thing. I don't boast—I don't join your friendly insinuation—that other composers who work with computers are less personal interpreters of the installation. The difference between some of them and me could be that I'm aware of that and they are not. There are people who really think they could not be personal. That's where the word and its misuse come from. It is a pointer to a liberty we do not have. We are always personal, no matter what we choose to say. Therefore we do not really have this liberty; just like an "own" opinion, "personal" is one of those "fakes" of common communicative language. It is a question of how conscious you are of having made a decision. This is the point. A composition, and even the first approach to what you want to do now, must be the result of a decision. This is not a law I lay down, but if you ask me about how I'm doing it, this is what I say. I consult my criteria and ask them what to do next. At that point I begin to ask for the indispensable and sufficient answer. Both are important: the answer must not leave out what is needed, and I should not, for alleged safety, add the superabundant; but it should be sufficient. So between those two poles I play my games. I think everybody is doing it, but with different degrees of awareness. Due to age and a very lucky situation in the teaching field—I've been kept alive an unduly long time—this particular awareness is still as good as new.

Hamlin: It's an interesting thing you brought up, because earlier you said you were trying to, in a sense, remove your person from what you were doing and now you're saying that's sort of impossible to do even with a computer, which is perhaps the most likely way to remove yourself from your person.

Brün: Yes. I was taking your terms more literally than you understood yourself. I replied that way not so much with regard to my person, but rather with regard to the inherited musical universe that harbors me instead of me harboring it, and which I like to make nonfunctional, at least as a decision criterion. I cannot wipe it out, nor do I want to. But I can appoint criteria for decision-making and I can also fire them. You can too. As a simple but understandable figure of the imagination, we each have in our minds a committee of "experts" which are the criteria we will consult when making decisions. These criteria are of various kinds: some are inherited, some are needs, but there are also appointed criteria, and there is a time in which they can and will be in this appointed position. If, however, you find repeatedly that this committee doesn't come to a conclusion you actually approve of, you fire it. But then you have to find other criteria. Composition is a wonderful method for discovering not-yet-appointed criteria.

Hamlin: It would seem to me that it would be hard to recognize those things occurring. Wouldn't it be easy to have your committee existing in a subconscious realm that you're not even aware of?

Brün: If there is a subconscious realm, I say simply: "Yes, it is possible" and so I prefer to say there's no such thing: that's talk. The subconscious, if it exists, is part of my profile—my "I" which is to make the decisions—and it is not a criterion.

Hamlin: Well, then that's probably not the proper word. How about awareness versus nonawareness? You could be aware of these things or not aware of them.

Brün: Here I discern between people who think what they say and those who say what they think. The majority are the people who think what they say because they have learned the sayings for twenty-five years. They then obey, due to these sayings we call *communicative language*, a package of thinking patterns which they cannot change anymore because they don't know the grammar and the syntax. The moment we stop learning how to speak, we have stopped thinking. Following the sentence structures that are called communicative language or "the language common to us all," we accusingly admonish one another: "Why don't you express yourself more simply?"—which is just another way of suppressing one another.

There's no hope for us to compose, to add something, to be an input to this society, unless we master the language more than the language masters us. There has to be this awareness of: "My god, what did language make me say right now" and "Why did I not succeed in making it say what I think?" These questions are continuously in the forefront. Composition is the way out of it. When I compose, I actually make a language say something that it would not have made me say. That's why I say I must not like it yet. It must not yet—I emphasize *yet*—have this communicative, cuddly appearance of "Oh yeah, I see what you mean." I simply *don't* want to see what I mean.

Hamlin: I must say that despite what you're saying about your own piece, I found it very likable. Is there something wrong with me?

Brün: Yes, there may be something wrong with you, if we can't iron out the word "wrong."

Hamlin: I mean I found the sounds very appealing. Perhaps it was because I've heard so many pretty, easy-to-listen-to sounds generated by computer.

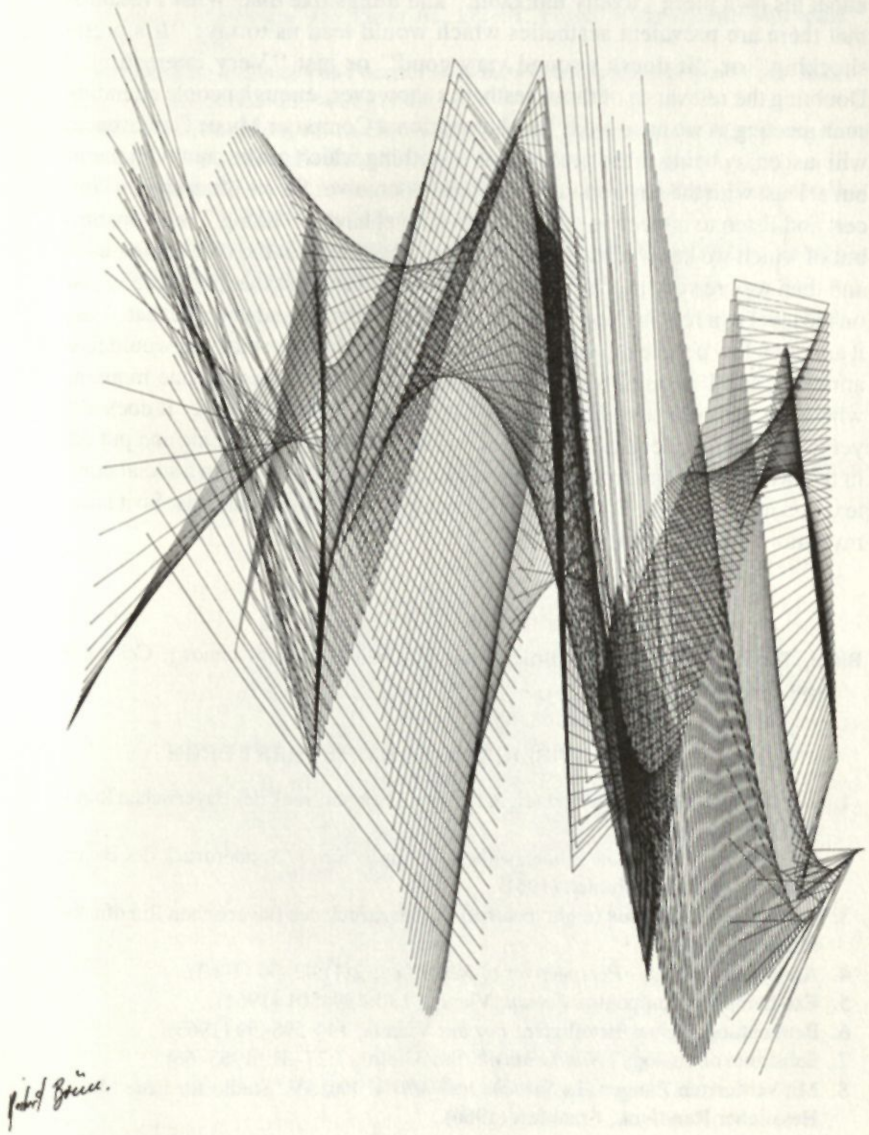


Fig. 1.2. Untitled computer graphic by Herbert Brün

Brün: I am aware of the danger of coquettishness when a composer says about his own piece "totally unlikable" and things like that. What I meant is that there are prevalent aesthetics which would lead us to say: "It's pretty shocking" or "It doesn't sound very good" or just "Very interesting." Doubting the relevance of those aesthetics, however, enough people attending such meeting as we have today [the International Computer Music Conference] will just enjoy being in the presence of something which grates, not with them, but at least with the environment which they perceive. So we often sit in a concert and listen to a piece to which we do not yet have a "liking" relationship, but of which we know already that it annoys the people in the row behind us—and then we are very much for that piece. I would suggest that my piece is just on the level where it invites you to a conspiracy with me, and you like that. Yes, it annoys a few people in your imagination or your presence that you would like annoyed, and I'm doing you this little favor. I provide you with one moment where that happens and then you like having been in that presence. It does not yet mean that the piece is one that you would voluntarily take home and put on in the evening to enjoy with a cigarette and a glass of wine. But in a social context you may have liked the fact that it happened rather than what it is. So it is an invitation to this conspiracy.

REFERENCE

Blum, Thom. 1979. "Herbert Brün: Project Sawdust" (record review). *Computer Music Journal* 3(1):6–7.

APPENDIX A: PUBLICATIONS BY HERBERT BRÜN

1. *Das Schaffen Gustav Mahlers* (eight lectures). Sonderdruck des Bayerischen Rundfunks, Munich (1960).
2. *Synthetischer Klang und Klangsynthese* (eight lectures). Sonderdruck des Bayerischen Rundfunks, Munich (1961).
3. *Musik und Information* (eight lectures). Sonderdruck des Bayerischen Rundfunks, Munich (1962).
4. Against plausibility. *Perspectives of New Music* 2(1):43–50 (1963).
5. Existieren als Komponist. *Forum*, Vienna, 130:499–501 (1964).
6. Betrachtungen eines Beteiligten. *Forum*, Vienna, 144:598–99 (1965).
7. Substitute or analogy? *Nudita Musik*, Stockholm, 7:27–31 (1965–66).
8. Mit verdorrten Zungen. In *Sprache und Musik*. Part IV. Studio für neue Musik, Hessischer Rundfunk, Frankfurt (1966).
9. Research on the conditions under which a system of computers would assist a composer in creating music of contemporary relevance and significance. In *Proceedings of 1st Meeting of American Society of University Composers*, 30–37. Princeton University (1966).

10. Chaos and organization. *Institute of Contemporary Arts Bulletin*, London, 166:8–11 (1967).
11. Muzyka i informacja. *Res Facta III*, 172–91. Polski Wydawnictwo Muzyczne, Krakow (1969).
12. Infraudibles. In Heinz Von Foerster and James Beauchamp, eds. New York: *Music by computers*, 117–21. Wiley (1969).
13. Mit verdorrten Zungen. In Ulrich Dibelius, ed. *Musik auf der Flucht vor sich selbst*, 45–54. Munich: Hanser Verlag (1969).
14. From musical ideas to computers and back. In Harry B. Lincoln, ed. *The computer and music*, 23–36. Ithaca, N.Y.: Cornell Univ. Press (1970).
15. *Über Musik und zum Computer*, 129 (including a record and 20 computer graphics by the author). Karlsruhe: G. Braun Verlag (1971).

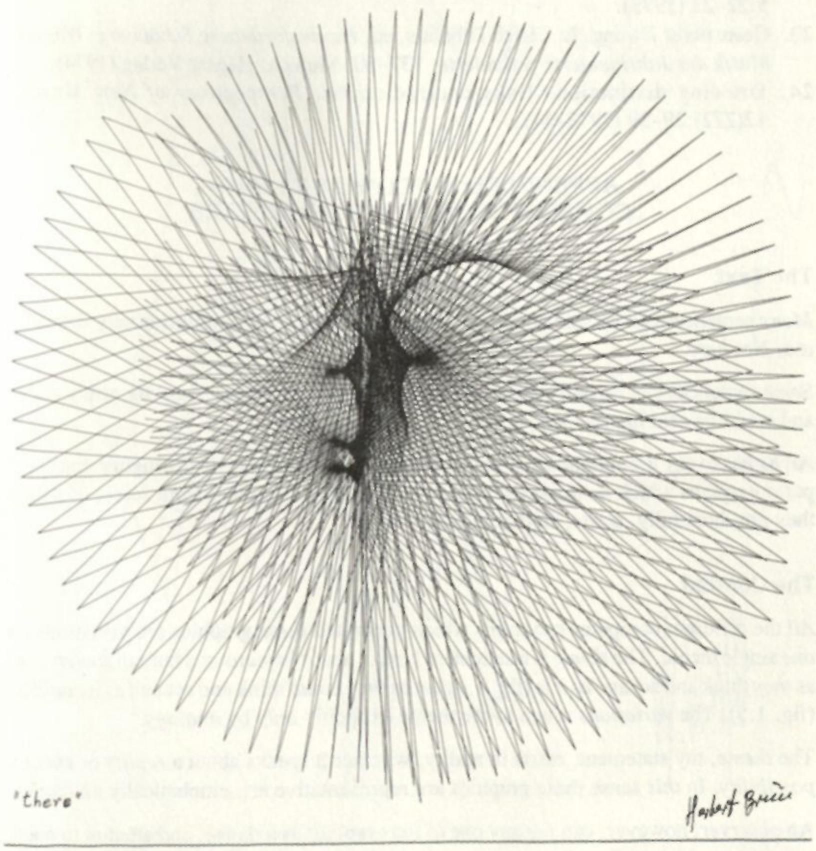


Fig. 1.3. "there": computer graphic by Herbert Brün

16. Technology and the composer. *La Revue Musicale*—UNESCO meeting: Music and Technology, 181–92 (special edition of papers presented in Stockholm, Sweden, June 1970), Paris (1971).
17. Technology and the composer (expanded version). In Heinz Von Foerster, ed. *Interpersonal relational networks*, 1/1–25. CIDOC Cuarderno No. 1014, Centro Intracultural de Documentation, Cuernavaca, Mexico (1971).
18. Mutatis mutandis. *Neue Musik*, 7–11 (special edition for the Olympic games) Munich (1972).
19. Probleme der Verständigung. *HiFiSterophonie*, 587–90. Karlsruhe: G. Braun Verlag (1973).
20. . . . to hold discourse, at least with a computer. . . . *Guildhall School of Music and Drama Review*, London, 16–21 (1973).
21. Mutatis mutandis. *Numus-West* 4:31–34. Mercer Island, Wash. (1973).
22. Mutatis mutandis: Compositions pour interprètes. *Les Cahiers Sesa*, Paris, 5:22–23 (1973).
23. Geste unter Zwang. In Ulrich Dibelius, ed. *Herausforderung Schönberg: Was die Musik des Jahrhunderts veränderte*, 137–50. Munich: Hanser Verlag (1974).
24. Drawing distinctions links contradictions. *Perspectives of New Music*, 12(272):29–39 (1973–74).

APPENDIX B: *MUTATIS MUTANDIS*—COMPOSITIONS FOR INTERPRETERS

The Text

Many sentences can be said about *all* computer graphics. *Will*, unfortunately, be said, too. Not *here*.

Some sentences can be said only about *some* computer graphics. They are rarely *found* and could *be* said *loudly*. *Not*, however, by the *composer*.

All *he* has to say is contained in a few statements indicating how to distinguish the computer graphics which *he* made from *all*, of which they are *some*, from *some*, of which they are, hopefully, *a few*, so that *they* be *these*.

The Context

All the different computer programs which generated *these* graphics are variations on *one* single theme. The *theme* is a statement I make about *humans* and human *society*, not as *they* think and act and as *it* is (fig. 1.4) but as they *could* think and act and as it *could* be (fig. 1.5). The *variations* relate to the theme explicitly *only* by *analogy*.

The *theme*, my statement, *exists* in reality, whether it speaks about a *reality* or about a *possibility*. In *this* sense these graphics are representative art, emphatically an *output*.

An *observer*, however, can *see* any one of the graphics as a *theme*, and attempt to make statements which *reflect*, by analogy *and* mutatis mutandis, the theme he *sees*. In *this* sense these graphics are, *until* the observer will have composed *his* statements, *non*representative art, emphatically an input.

The Theme

As long as we do not abandon *present* society, *future* society is "anarchy and chaos."

In a desirable society which, as we are *not* it, is a future society, each of us, its members, moves through life along some path composed of *steps taken* in preference to many equally possible and equally desired *steps not taken*. The preference is with each of us,

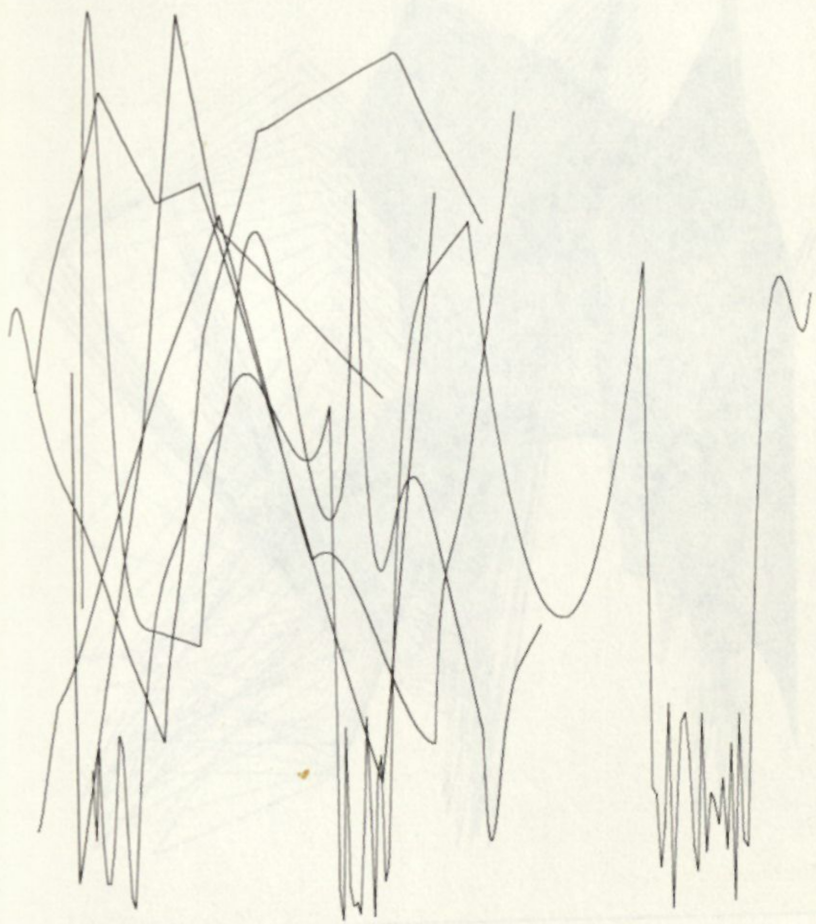


Fig. 1.4. 9069-5b: computer graphic by Herbert Brün

each member. It is directed, however, not by *each* of us contemplating *his* desired path (fig. 1.4), but rather by *all* of us contemplating the contribution of *every* step of *every* member to formations of relations (fig. 1.5). A step is preferred when found, beyond being desired, to also be desirable.

Unless we *abandon* present society, future society will *be* anarchy and chaos.

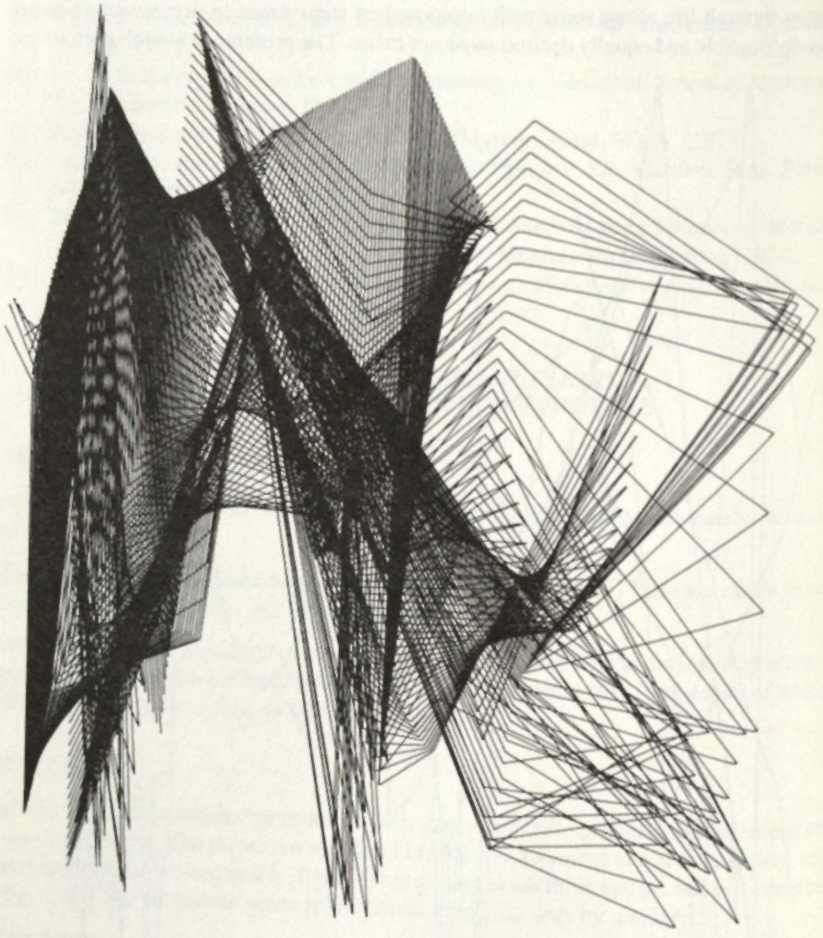
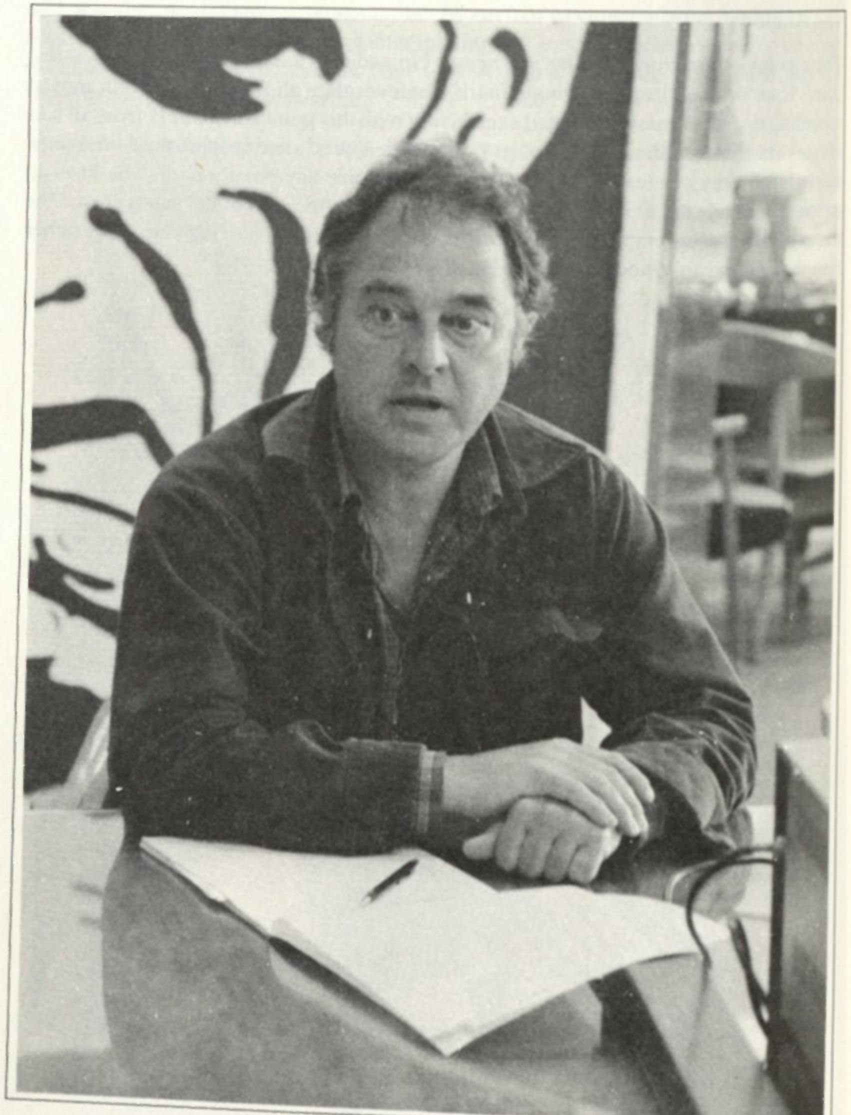


Fig. 1.5. 9069-5a: computer graphic by Herbert Brün

The Analogy

These graphics are *traces* left by a *process.* Up and down and across the page a few points leapt in small leaps, leaving a mark wherever they alighted. Each point moved according to a set of rules associated exclusively with this point and *distinct* from all sets of rules associated with any of the other points. The lines do *not* connect the *consecutive* marks left by any *one* leaping point, they do not outline any point's *path.* The lines *do* connect; instead, the new marks left by *every* point after *all* points' latest leap. The traces of *this* process emphasize the *shape* created by *all* points *moving* (fig. 1.4), rather than the *outlines* followed by *each* point's *leap* (fig. 1.5).



JOHN CHOWNING

2

JOHN CHOWNING ON COMPOSITION

Curtis Roads

John Chowning (b. 1934, Salem, New Jersey) is the founder and director of the Center for Computer Research in Music and Acoustics (CCRMA) at Stanford University in California. This interview took place the afternoon of 29 April 1982 at CCRMA.

BACKGROUND

Roads: Could you tell us about your early musical experiences and education?

Chowning: My background is thoroughly traditional. I started playing violin as a child, and later I played percussion instruments in my teens. I became interested in jazz in high school. I then went to the Navy School of Music for three years, during the Korean War. I had a lot of exposure to some awfully good musicians—people like Nat and Cannonball Adderly. There was a lot of good jazz activity.

Then after the Navy I went to college. That's when I became interested in composition. I improvised a lot as a percussionist and became more and more interested in composing. Following college I studied with Nadia Boulanger in Paris for three years, from 1959 to 1962.

Roads: Was there anything about the European musical scene at that time that especially interested you as a composer?

Chowning: The electronic music. That was a very active time in Paris. Pierre Boulez had the Domaine Musicale concert series going. I heard all the current performances of important composers being done there, like Stockhausen's *Kontakte*, Berio's *Circles*, and new pieces by Haubenstock-Ramati and Henri Pousseur, for example. So it was really lively—quite in contradiction to the Boulanger environment. In fact, that wore me out, I must say. After about a year and a half I was ready to stop. The third year I just wrote music and participated in the concerts.

Then I came to Stanford, where I was to do my graduate work. Largely as a result of my exposure to electronic music in Paris, I inquired about the possibility of electronic music here. There was no studio—and certainly no interest. However, they did have a rather good computer for the time, an IBM 7090. This

was a great big machine in those days. It shared a disk with a DEC PDP-1. It was the beginning of the Artificial Intelligence Project here with John McCarthy, who had come from MIT in 1962.

So, with the help of David Poole and by the courtesy of McCarthy, we got Max Mathews' program Music IV going on the 7090. The sample data was written onto the shared disk, and we used the PDP-1 as a kind of buffer to the *x-y* digital-to-analog converters on the DECscope [a display terminal] for sound output. The first sound we made was in September of 1964.

Roads: How did your musical background affect your later compositional thinking?

Chowning: The rigorous education one gets in music, such as harmony and counterpoint, is still an important part of the way I think—especially counterpoint. I agree with Luciano Berio in that I believe the study of counterpoint pays off. There's probably no other way to gain an insight into the working of musical lines like going through species counterpoint. That's very much a part of me despite the fact that computers figure most prominently in my musical world today.

Improvisation also affects me deeply. The freedom one has in improvisation seems opposite to the rigor of counterpoint.

SOUND IN SPACE

Roads: When did you begin your research into the computer-controlled movement of sounds in space?

Chowning: That was my first project in 1964 when I started. It came from thoughts that were common in contemporary music at the time. There was plenty of electronic music in Europe at the time which attempted to utilize space in a fairly primitive way. Nevertheless, the idea was there.

Some of the computer research I did was obvious and some was not. The obvious work involved using multiple channels of sound to build up an image of a source at some arbitrary angle with respect to the listener. The question of distance, and the relationship of distance to reverberation, was not well understood at that time. I think that research was more interesting, and we are only beginning to realize the consequences of it. I can talk a little bit about that in a moment. The use of Doppler shift was a natural consequence of moving a sound at an angle over some distance.

Roads: Could you explain Doppler shift for the benefit of our readers?

Chowning: Doppler shift is the change in frequency that occurs when a sound source is moving toward or away from the listener. If I have a buzzer on a string and I'm twirling it over my head, I don't hear any Doppler shift. This is because there is no change in relation to my position; there is a constant radius. But you, the listener, standing near the perimeter of the buzzer's trajectory, will hear a pronounced Doppler shift. The sound will increase in frequency as it comes toward you and decrease as it goes away. In any case, it's a cue to the motion of sound in space—in particular, to the radial velocity of a sound, as opposed to angular velocity. So what I did was write a program that incorporated a distance cue, an angular cue, and a velocity, in such a way that a composer could use it gesturally. A composer could specify geometrical sound paths in a two- or three-dimensional space (Chowning 1971).

Roads: In which compositions did you use these spatial programs?

Chowning: In *Turenas* I made extensive use of these programs. I also used them in my first computer piece *Sabelithe* (1971). *Turenas*, which is a four-channel composition, was probably the most effective use.

TURENAS AND FM

Roads: When was *Turenas* composed?

Chowning: It was completed in the spring of 1972. The composition work spanned several years, however. I was involved with writing the spatial manipulation programs for some time, and *Turenas* made extensive use of that experimentation. It's hard to say when a composition begins if research is tied so intrinsically to a work. The piece evolved over a period of years, and I finally finished it after I concluded that I had enough of the music-gestural control over the computer.

Roads: *Turenas* is based on the *frequency modulation (FM)* sound synthesis technique, a technique based on your own research. How is FM used in *Turenas*?

Chowning: FM is something I stumbled upon in the mid-1960s. It turned out that one could, in a sense, "cheat on nature." By modulating the frequency of one oscillator (the *carrier*) by means of another oscillator (the *modulator*), one can generate a spectrum that has considerably more components than would be provided by either of the two alone (Chowning 1973).

There's another important aspect. FM provides a simple way to get dynamic control of the spectrum, which is one of the aspects of natural sounds that was very difficult to reproduce with analog synthesizers. So FM is a synthesis technique that is useful or not depending upon the type of control one desires. It turns out to be quite widely used, and its usefulness is that it provides a few handles onto a large timbral space.

In *Turenas*, I used only the FM technique for generating the tones. I used it in both a *harmonic series* mode and a noisy *inharmonic series* mode, with transformations between the two. One of the compositional uses of FM was in timbral transformation. This was often coupled with spatial manipulation. As the sounds crossed the space they underwent a timbral transformation.

Roads: How was this accomplished?

Chowning: There were a number of techniques. Sometimes there were very slow transformations from harmonic series timbres to other harmonic series timbres—from rich double-reedy sounds to flutelike sounds. In that case, there was a gradual change in modulation index. Other kinds of transformations in the piece had to do with changes from harmonic to inharmonic spectra or the inverse, through a gradual change in the *carrier-to-modulator* (*c:m*) ratio.

STRIA

Roads: Would you say there's a kind of dualism in your music based on competing tendencies towards rigor and improvisation?

Chowning: Yes. *Stria* (1978) was rigorously composed. *Turenas* was much more improvisatory. They both feel natural to me. *Stria* was probably the most fun piece I have ever composed.

Roads: That was rigorous composition.

Chowning: Right. I just got into it. It was the first time I'd tried to use a high-level programming language to realize a composition in toto. I learned a lot and I enjoyed the rigor of it all. Then at some point it became magical when it was all working!

Roads: How was *Stria* organized?

Chowning: It was based on an idea that occurred in the early 1970s. Just after I'd finished *Turenas* I was doing some experiments with FM synthesis using inharmonic spectra. I marveled at the fact that in setting inharmonic ratios between carriers and modulators, that unlike in nature, there was a perceptible order when one moved through the frequency space with a constant spectrum. Even when I changed the envelopes, there seemed to be something remaining that was certainly distinct from the harmonic series but was still ordered.

Then when I was in Berlin in 1974 and had no computer to use, but had lots of time, I thought about all this. I was looking for an inharmonic ratio such that the components would be powers of some basic ratio. It turns out that the

Golden Mean (1.608) is such a number. If one has a *c:m* ratio that is 1 to some power of the Golden Mean, then several of the low-order spectral side components are also powers of the Golden Mean.

What I did was draw an analogy between this inharmonic spectrum—including a frequency space where the pseudo-octave is at powers of the Golden Mean—and the harmonic series and tonality, where the low-order components of the harmonic series are also the principal intervals of the tonal system—the octave, the fifth, and so on. I drew this loose analogy and wrote some programs to help me compose, in particular to help me with the sound synthesis. It was not automatic composition by any means, but there were rules for determining the details of the structure, from the microsound level up to the level of a phrase.

In *Stria*, all frequency components are based on powers of the Golden Mean in the *c:m* ratios. Then I divided up the frequency space so there was some degree of complementarity. So it is all very cohesive perceptually, even though it's inharmonic and sounds a little strange. But it doesn't take long, even for a naive listener, to realize that even though it's strange it's cohesive at a deep level. I believe this is because of the unified structure of spectral formation.

SYNTHESIS OF THE SINGING VOICE AND *PHŌNĒ*

Roads: When did you go to IRCAM, the French musical research institute?

Chowning: I was associated with some of the plans at a developmental stage in the mid-1970s. I made some of my thoughts known about interesting directions. Others from CCRMA, including Andy Moorer, John Grey, and Loren Rush, were also involved. Then I went there for about eleven months in 1979 and 1980. I developed some algorithms based on FM for synthesis of sung vocal tones (Chowning 1980).

Roads: You used these tones in your composition *Phōnē*.

Chowning: That's right. *Phōnē* is based exclusively on the use of this algorithm. The idea was inspired by some work of Michael McNabb's here on the additive synthesis of sung vocal tones. I hadn't intended to work on that when I went to IRCAM, but I took it on in order to familiarize myself with their system. It turned out that Johan Sundberg was there at the time, a wonderful scientist from Sweden. He has done considerable work in the analysis of the singing voice. So I had this tremendous resource at my elbow, and I was seduced by the problem. I became extraordinarily interested in naturalness. I found that all the previous attempts at vocal synthesis really lacked something. So I developed this algorithm and tried to embed in it as many performance characteristics as I could. This meant understanding them. For example, how

much randomness in periodic vibrato must be present in order to create a convincing impression? Or, must a sung vocal tone have a little portamento in the attack? Or, how do the formants behave during the attack and decay portion of a sung vowel? It turns out that all these things are very important. My stay at IRCAM could be characterized as "tending to detail."

Having done all this, I found that interesting ambiguities occurred if there was neither periodic nor random microfrequency variation. One can make sounds that sound like an instrument and then evolve into vocal-like tones.

Roads: Where does *Phōnē* stand on the scale of rigorous organization versus improvisation?

Chowning: Right in the middle. I also used computer programs to control the low-level synthesis as in *Stria*, but I think there was more fantasy in its composition.

THE SOURCE OF COMPOSITIONAL IDEAS

Roads: Where do your compositional ideas come from? Do they come from imagining large-scale structures or processes, or do they come from within the sounds themselves?

Chowning: They come from several sources. Certainly all the time that I and others have spent over the years looking at the internal workings of sound at the microstructural level has influenced the way we proceed. This is something that in traditional composition one doesn't normally do. There is no doubt that *Stria* evolved from a microstructural notion. The piece as a whole reflects the shape of the event in its smallest unit.

But I must say I get a great deal of inspiration from computer programming languages. The idea of a procedural language reeks of music somehow. I've just barely touched that domain. It's clear to me from watching others work in this lab, using programs like Bill Schottstaedt's Pla program, that computer languages are extraordinary resources.

Most of the music being written here at CCRMA involves powerful algorithmic processes. It is very different from the note-by-note Music V kind of input to the computer. These algorithmic approaches are obviously rich because they are being used so widely and the music is so good.

The language is important. It is a lot easier to do things in a modern high-level language than it was with FORTRAN or assembler, for example. More and more, the musical idea evolves from a kind of cyclical interaction with the language. One asks something of the language and it yields more than you asked for. That's not surprising since the language represents thousands of years of thought about thought.

Then, of course, fantasy is another component of my compositional thinking. I can't talk very much about that because I do not know how to talk about it.

PERFORMANCE OF COMPUTER MUSIC

Roads: Many of the compositions produced at Stanford, including your own works, are a part of what could be called tape music, in that they are recorded and performance is a matter of playing back a tape. The Italian composer Luciano Berio stated in a recent interview that he believes tape music is dead (Schrader 1982). Would you agree with this assessment?

Chowning: No, I don't believe that's true. Would he say that any music that doesn't emanate from a performer is dead? That's clearly not true. There is an entire record industry that proves it is not. The question of whether a musical gesture is made in real time or not seems moot when one listens to a recording. After all, Berio's music is recorded, and most of us would not hear his music but for recordings.

To go into a space and listen to a tape or, as we can do here, listen to direct digital synthesis (which is the same idea, in that there is no performer) is clearly something people like to do. It is an important experience because one shares the event in a communal way that is deeper as a result of being shared. If you and I are listening to music together, there is surely something we are going to communicate about after the experience. It doesn't really matter whether the music has been performed by instrumentalists in real time or by the recorded sound of a composer controlling a computer.

There is another aspect to this, one which we play upon to good effect here at Stanford. One can present music in a concert situation in a manner it cannot be presented at home, using very fine audio equipment in a carefully planned context. We do outdoor concerts. Our audience is growing to the point where we now attract several hundred people to a concert. Well now, that's rather extraordinary for nothing but "tape music." I think the proof of the matter is that it can't be dead if it's alive and well at Stanford and a number of other places.

Sure, we would all like to have more performance involved. I don't think any of us who are working in the medium feel that performance is to be excluded—quite the contrary. For years and years we have wished that digital systems were cheaper and smaller such that we could introduce the performer into the complex. We hope it will happen—and soon—but it's not exclusive, it's additive—another use of the computer.

FUTURE COMPOSITIONAL DIRECTION

Roads: Are you working on a composition at the present time?

Chowning: Well, I am trying to. I am also administrating CCRMA, and these last months it's taken a lot of time for fund-raising. But the ideas are there for a piece to be realized at the moment I get enough time. It reverts back to my early work in spatial processing, but having rethought what all that means in a more musically important way.

The issue of distance is one that has much more importance than simple questions of sounds being close or far. It impacts the whole notion of loudness. Loudness has been used in a rather unsubtle way in electronic music in general. Pressure level has been seen as the measure of loudness, and if the composers want something to be louder, they up the gain, by computer or potentiometer. That is not what loudness is; that's only part of it. The richness of the loudness space correlates with the richness of the visual space. I believe there is an analogy to perspective in vision, and that is auditory perspective. The piece I am thinking about is going to exploit this notion in a rather deep way.

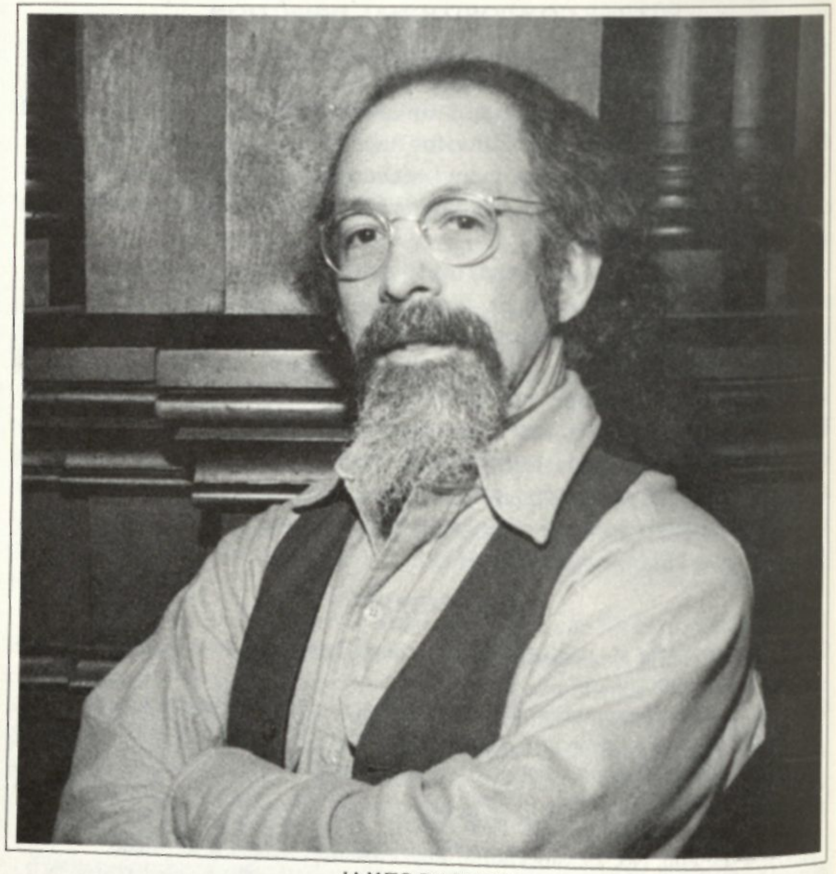
REFERENCES

Chowning, J. 1971. The simulation of moving sound sources. *Journal of the Audio Engineering Society* 19(1):2–6.

Chowning, J. 1973. The synthesis of complex tones by means of frequency modulation. *Journal of the Audio Engineering Society* 21(7):526–34. Reprinted in C. Roads and J. Strawn, eds. 1985. *Foundations of computer music*. Cambridge: MIT Press.

Chowning, J. 1980. Synthesis of the singing voice by frequency modulation. In E. Jansson and J. Sundberg, eds. 1980. *Sound generation in winds, strings, and computers*, 4–13. Pub. No. 29. Stockholm: Royal Swedish Academy of Music.

Schrader, B. 1982. *Introduction to electro-acoustic music*. Englewood Cliffs, N.J.: Prentice-Hall.



JAMES DASHOW

3

INTERVIEW WITH JAMES DASHOW

Curtis Roads

INTRODUCTION

James Dashow was born in Chicago in 1944 and has lived in Rome since 1970. In the spring of 1983, he was a visiting professor at the Massachusetts Institute of Technology, associated with the Experimental Music Studio. This interview was recorded in March and June of 1983 in Cambridge, Massachusetts.

BACKGROUND

Roads: Could you tell us about your composition background?

Dashow: I did my undergraduate studies at Princeton University between 1962 and 1966. From the very first day I had contact with J. K. Randall, who was one of the first people to be turning out finished and serious compositions in the computer medium. I didn't know it at the time, but Randall was already rather heavily involved in some of the key problems that most of us face when we enter computer music. I also worked with Earl Kim and Milton Babbitt in composition and with Edward T. Cone in theory and composition.

Roads: Which of your Princeton teachers had the most influence on you?

Dashow: Edward Cone introduced me to the tradition of Western music. But from the standpoint of a composer working within the field and tackling the problems of creating music, I think J. K. Randall was the most important influence. His output of music is not enormous, but his extremely open attitude toward all kinds of music was something that had a great effect on me at that time.

Roads: Didn't you have some background playing jazz before you came to Princeton?

Dashow: That's correct. I was an alto saxophonist in high school. This meant gigging around in the Chicago area with various combinations of bands, ranging from Ornette Coleman-style trios all the way up to big bands. At one point I had a 17-piece jazz band. We managed to actually find some work every now and then. I did arranging for this band, and my first real compositional efforts were also for this group.

Roads: What was your first major composition?

Dashow: The first extended composition that I took rather seriously was a large piece for my 17-piece jazz band, for 5 saxophones, 8 brass, and a rhythm section. This was in 1961. As far as a composition that I would present today on a concert, that would be *Songs of Despair* (1968–69), for soprano and 11 players, or *Timespace Extensions* (1969).

ELECTRONIC MUSIC

Roads: When did you become interested in the possibilities of electronic music?

Dashow: My first real contact with electronic music was in graduate school. Brandeis Universty had a small analog studio, nothing very sophisticated, with a few tube oscillators and a couple of Ampex 350 tape recorders. By today's standards their studio would be next to nothing. But I remember one day puttering around in there, and I began to hear musical possibilities. I began to put together little electronic music etudes.

Roads: Who did you study with at Brandeis?

Dashow: I worked with Arthur Berger and Seymour Shifrin who, along with J. K. Randall, were the strongest influences on the way I think about music and the way I hear music. Arthur was beginning to write some piano pieces around 1968 which became the *Five Piano Pieces*. They struck me as being extraordinarily "right." That composition had an enormous effect on me. It is still a kind of sound that I try to capture, in my own way.

Roads: What was that sound?

Dashow: Arthur uses lots of minor second dyads. He spreads these out all over the piano to make minor ninths and major sevenths and combinations of these dyads at various transposition levels. It makes a very transparent and open kind of articulation on the piano. This openness struck me as being the most persuasive part about it.

I was attracted not only to the style of writing but also the way Arthur used the timbre of the piano to achieve a kind of clarity and ongoing complexity in the music. It was an almost subterranean sort of complexity since it had a lovely surface, but when you listen to it more closely you hear very complex but clear musical ideas. It is a piece I have listened to many times and still find fascinating.

CLARITY, OPENNESS, COMPLEXITY

Roads: I heard three adjectives: clarity, openness, and complexity. Would you say that those three properties can be found in your own work?

Dashow: Yes, I would say that those are my goals. I attempt what might seem paradoxical, being complex and yet open, or complex and yet clear. I enjoy complexity in music. It challenges me, and, when I feel I have come to grips with a well-made piece of this sort, I derive pleasure out of the sense of having met and understood a fine mind.

Complexity is a social issue, too. This "global village" we live in is an extremely complex affair, and people must be aware of this complexity and deal with it in responsible ways. That is to say, accepting simplistic, facile solutions to world social problems is to ignore this complexity and behave irresponsibly. Complexity in art is one way of reflecting this situation, calling it to our emotional attention in meaningful ways. It encourages our emotions and intelligence to understand, or at least not be afraid of complexity in life. The world faces some very dangerous problems, and facile solutions based on simple slogans don't work. We have to be prepared to confront the frighteningly complex issues that surround us.

This is why I am unhappy about the proliferation of certain kinds of banal commercial music, and why I am becoming increasingly alarmed by the inclusion of such music in works by composers who would claim a certain seriousness of intent. Besides being a sad reflection of musical values and taste, this trend is indicative of a sort of creeping mindlessness that is everywhere in our society. This mindlessness encourages the acceptance of dangerously simplistic answers to the complex problems of our time.

I have been told in various ways by many well-meaning people that my music should be more realistic and reflect popular taste and so reflect society. But this is wrong. Reality is complexity, not simplicity. It's the commercial music and the message it conveys that is unreal, promoting a society unresponsive to anything but consumerism. Commercial music is unacceptable as a substitute for music that attempts to challenge, prepare, and, I hope, satisfy a thinking, responsible society. Perhaps I have gotten carried away, but I am rather involved with this issue.

EARLY WORKS

Roads: During what period were you at Brandeis?

Dashow: Brandeis was immediately after Princeton, from 1966 to 1969.

Roads: So it was in those years that you wrote *Songs of Despair*?

Dashow: That's correct. I started *Songs of Despair* and interrupted it because I had a request from Max Polikoff in New York for *Timespace Extensions*. He wanted the piece for an April 1969 concert, my first New York performance. I dropped work on *Songs of Despair* and got to work on *Timespace Extensions*. So those pieces are simultaneously my "opus one."

STUDY IN ITALY

Roads: What about the period after Brandeis?

Dashow: During my last year at Brandeis, I had the good fortune of winning a Fulbright fellowship to study at the Accademia Nazionale di Santa Cecilia in Rome with Goffredo Petrassi. I went, and that turned into my unfinished Italian career, as it were.

Roads: What was it like to study with this well-known figure in Italian music, Petrassi?

Dashow: Petrassi was almost two different people. My first year was like a master class. There were five or six of us from different parts of the world. Some were second-year students, and some were first-year students. Petrassi was not in good humor. He seemed unable to make instructive suggestions, though he was obviously an intelligent and sensitive person.

Someone told me later that Petrassi had cataracts on his eyes and that was the reason for his very bad humor. During the summer between my first and second year with him, he had an operation, and the next year everything cleared up. The man looked like he had lost ten years in the process of one year. He was full of good spirits; he was warm and outgoing. He readily admitted that he was not attempting to convert us to his way of thought musically. He was there to give us suggestions about what he thought we were doing in our music. For the second-year course there were only two of us—me and a Spanish composer, Jesus Villa-Rojo. He has since become the head of a well-known group doing electroacoustic music in Madrid: Laboratorio de Interpretacion Musical. Petrassi was an inspiration as a human being, though he would often look at one of my compositions and murmur, "Well, I don't know what I can say here."

He did make one comment to me which turned out to be perhaps the single most valuable comment that any of my teachers has ever said to me. He said, "Dashow, this piece needs more space," and for some reason I understood exactly what he meant that day. I imagined a sense for space, the kind of space that Arthur Berger realized in his piano music. I finally knew what Arthur was doing both from a technical point of view as well as the gut level. There was something about that day and the way Petrassi said it that made it all click—it all made a great deal of sense to me.

One day Petrassi walked in and didn't feel like teaching. He began telling me about his experiences in New York, and how he thought American men were persecuted by their wives, and so on. Then he pulled out a book with which he thought he was going to shock me. It was a paperback edition of the last speeches of Lenin, and he said, "Dashow, I think you should read this." From subsequent conversations I've had with him, it was clear that Petrassi, like many Europeans, has an image of the United States as a politically naive country without access to political ideas other than those of the establishment. While there may be some truth to that, I wonder if as many Italians are as well versed in, say, the Federalist papers, as their American colleagues are in Marxist-Leninist writings.

POLITICS AND MUSIC

Roads: Have political ideas found their way into your compositions?

Dashow: I doubt it. However, Italy is a very politically conscious nation. You literally cannot even buy your pasta without someone deciding that you are somehow politically motivated. I have been forced to be aware of the fact that what I am doing does participate in our Western society, that it does represent a certain level of cultural and technical knowledge. We come back to the old question about whether art has any validity in a complex society where people are starving. One should be aware of those things as a responsible member of society; on the other hand, I don't think music necessarily has to represent those things. I remember Randall saying he wasn't the type to face the cops at the barricade, but his pocketbook was open to those who would.

I certainly do not feel the need to do any kind of social sermonizing with my music. In the absence of specific words or extramusical "dedications" to some cause or person, I don't think music can be political. Some of my musical colleagues in Italy who are members of the Communist party would perhaps accuse me of being irresponsible as an artist. The composer as a member of society, as a person, does have political and social responsibilities, especially when society is rotten. But as a composer, my responsibility is to my art, to making music. Let composers make their music and support the cause with the prestige of their accomplishments, but don't let the politics of the cause support the making of the music. Mozart is played everywhere.

MUSIC IN ITALY AND THE UNITED STATES

Roads: What would you say are the differences between the musical scene in the United States and the musical scene in Italy right now?

Dashow: That's a complex question. With the ease of communication between all parts of the world, there is always cross-fertilization between

developments in Europe and in the United States, although many Europeans would not admit that they are being fertilized by their brothers and sisters across the ocean.

One of the main differences is the quality of contemporary music performance. The United States has two or three centers of extremely high-quality contemporary music performance. New York and Boston, in particular, are two areas that have a good deal to be proud of. There are other groups in the U.S. where the sense of dedication and professionalism to contemporary music is quite astounding.

By contrast, in Italy the degree of professionalism in contemporary music is shockingly bad. Conservatories don't prepare the students for contemporary music. Voice teachers tell their students not to sing contemporary music because it ruins the voice. The musicians are totally unprepared to do new music, and, as a result, they do not develop any kind of allegiance or love for it. Why should they, when most of the paying work is with old music anyway?

Roads: Was it your concern about this situation that led you to form your own performance group in Rome?

Dashow: Yes. It seemed when I arrived in Rome in 1969 there were only two groups that were performing contemporary music. I discovered that there was a vast pool of foreign musicians who had come to Rome for their master classes and who knew how to play contemporary music. They were looking for opportunities to play. So I began putting together a mixed bag of performers that consisted of young professionals from literally all over the world. I had people from not only the United States and Italy, but also Great Britain, France, Germany, Japan, Austria, eastern European countries, and the Soviet Union. The first year we did five concerts. The second year we did eight concerts, and four of us went on a Mediterranean tour to do a program of American contemporary music.

Roads: What was the name of that group?

Dashow: The Forum Players. We lasted until 1975 and then it became clear with economic problems it was impossible to carry on. Most Italian and European groups were able to come to Italy or play around Italy for next to nothing, since they all had financing by either the region they came from or by the federal government. We were a freelance group and we were not eligible for such financing, so we had to ask straight fees. Anybody who can get a group for free wasn't about to pay 500 to 1000 dollars for a group that would do more or less the same repertory as the free group.

ASHBERY SETTING

Roads: So, between the time when you arrived in Italy in 1969 and 1975 when the Forum Players were disbanded, you composed a number of instrumental works. Could you tell us about that compositional period?

Dashow: The first piece done specifically for the Forum Players was *Ashbery Setting*, for soprano, flute, and piano (1971-72).

Roads: Was this your first composition that utilized the poetry of John Ashbery?

Dashow: Yes. This was the first time I used an Ashbery text. I first discovered Ashbery's poetry while browsing in a bookstore in Harvard Square in 1968. I read repeatedly the two or three books that were then available, because I found something about his way of dealing with fleeting poetic images analogous to the way I conceived contrapuntal musical ideas.

He would suggest some sort of image and then would slide into some other image. You wondered how he got from one to the other. You always maintained in your mind the first image while he led you into the second, unexpected image. There was a sort of contrapuntal sense to these images, even though you read them one after the other. Images that were seemingly unrelated in their linear order became beautiful if you made them simultaneous in your memory.

This moving around among strata of freely associated images struck me as being similar to my conception of contrapuntal music, while maintaining that clarity and space which I had absorbed from Berger and Petrassi.

INTRODUCTION TO COMPUTER SOUND

Roads: At what point did you become introduced to working with computer-generated sound?

Dashow: I had my first touch of computers while I was still at Brandeis. The chairman of the music department at that time, Harold Shapero, was not sure whether he wanted Brandeis to get involved with analog electronic equipment or computer-generated sound. So in the spring of 1967, I was sent down to take the two-week course being given by Hubert Howe at New York University on the fundamentals of computer sound synthesis. He was using the program known as Music 4B, a version of Max Mathews' Music IV program written in a Bell Laboratories' dialect of IBM assembly language called BEFAP.

Howe was working at the time with Godfrey Winham creating Music 4BF, so it was a double course. Those who had access to an IBM 7094 could get a monitor tape and run Music 4B, or you could run the more laborious Music 4BF.

After two weeks of that, I stopped off at Princeton and picked up Music 4BF from Godfrey.

I remember coming back to Brandeis with four or five boxes weighing down my suitcase with punch cards and that was Music 4BF. I arrived at Brandeis and I said "Look here it is! Here's the monitor tape. Let's go over to the Harvard Computer Center and see what we can do!" Unfortunately Harold Shapero had decided to go for analog equipment. I will never forget how he looked at me and said: "What? You want to punch cards for the rest of your life?" So that was it for computer music at Brandeis.

COMPUTER MUSIC IN PADUA

Roads: You encountered computer music again in Italy, did you not?

Dashow: I was very enthusiastic about working with computers, but I was at a dead end. When I got to Italy I was doing nothing. I had dropped out of the digital field for what I thought was going to be a good long time, if not forever. In 1974 I met some of the people from the University of Padua who had just begun to do musical score encoding. They were interested in digital synthesis, but they didn't have the programs to do the job.

Since Hubert Howe was a good friend of mine, I wrote him a letter saying, "Please send Music 4BF!" Music 4BF showed up by return mail, and within less than a couple of weeks we had it up and running! It worked like a charm.

That was my first encounter with the extraordinary Giuliano Tisato, who wrote the interface programs that allowed us to use a prehistoric IBM System/7 as buffer to the digital-to-analog converters.

Roads: What year was this?

Dashow: 1975. It was at that time that I began to make my first plans for *Effetti Collaterali* (1976), which is my first piece using a computer as an accompaniment to a live performer. I realized the piece in 1976, but I had preceded that with about seven or eight months of preparatory studies.

COMPOSING FOR INSTRUMENTS AND COMPUTER SOUND

Roads: *Effetti Collaterali* is a piece for clarinet and tape. Could you describe some of the compositional considerations that went into that piece?

Dashow: *Effetti Collaterali* is the first piece that makes use of my notion of harmonizing dyads or triads of pitches with frequency-modulation (FM) spectra. I use the word *triad* in its general sense, not in terms of tonal triads or diatonic triads, but any collection of three pitches. What I am particularly interested in is the so-called *harmonizing* of these pitches with inharmonic frequencies.

Roads: Have you done pieces for other instruments and digital sound?

Dashow: Yes, two for voice, *A Way of Staying* (soprano) and *Second Voyage* (tenor), and the recently completed *Mnemonics* for violin and tape. In each case, the nature of the live instruments has made me develop different groups of sounds in order to achieve the best possible timbral blend.

Roads: What for you is the difference between composing for traditional instruments and composing for computer sound?

Dashow: The obvious difference is the timbral manipulations one can do with a computer. This is impossible to do with traditional instruments. Also, the rhythmic possibilities with a computer allow me to write infinitely more varied and complex structures than I would dare write for human performers, especially if I want precision.

The nature of the musical ideas that I write either for computer realization or acoustic instrumental realization generally follows the same basic idea. This involves relating predetermined groups of notes by common tones where the groups are the generating background for the timbral structure which I manipulate in various ways.

One of the most powerful uses of the computer is the generation of timbres that make systematic use of inharmonic relationships between the frequency components in a given sound. If you ask a human performer to play successfully a series of finely graded microtones, it will almost never happen. The computer has no problems with this task. These sounds are fascinating. They are the kind of sounds that I had begun to develop even with my analog electronic music.

With my generating dyad or triad technique, I am able to develop meaningful controls over the components of the spectra. In the classical FM technique that John Chowning or Barry Truax uses, a particular pitch or frequency is determined and is treated as the carrier or the modulator. In my technique, my selected pitches define two components of the spectrum, and I let the computer calculate the carrier and modulator frequencies to yield the notes I want in the spectrum.

I know from experience that any particular interval can generate for me a certain small group of distinctly different sound spectra. These have become the basic elements of my sound. Within a traditional orchestra, you know the characteristic sounds and pitch-interval capabilities of the instruments, so you compose for those notes and intervals. I do exactly the same thing with my computer orchestra, but in this case, some of my instruments are these modulation spectra.

Roads: *Conditional Assemblies* was also realized using this theory that you have developed for digital sound synthesis, was it not?

Dashow: Yes. *Conditional Assemblies* (1980) is a much more rigorous application of this particular idea. By the time I arrived at *Conditional Assemblies*, I had completed a major work, *Second Voyage* (1977-79) for tenor voice and electronics (available on CRI SD456). During *Second Voyage* and even more so for *Conditional Assemblies*, I developed other signal-generating algorithms.

In these, I made use of the fundamental idea of starting with two or three pitches and working backwards. I wanted to discover which frequency relationships between two signals that modulate or multiply each other would be necessary in order to realize a series of spectra with these pitches contained within them. These spectra are considered as the chords that accompany or "harmonize" the specific pitches that generated them.

I also applied this technique to ring modulation, amplitude modulation, and a controlled use of the foldover phenomenon. I then applied it to an algorithm based on the use of an exponential function as the signal generator. This was very uneconomic from the standpoint of computer time, but it made some interesting sounds. These were all developed in the *Conditional Assemblies* orchestra, which was realized with the Music 360 program at Padua, which we had been using since 1978. We made many modifications to the original MIT Music 360 program, and also added several special-purpose unit generators to its basic library.

ORGANIZATION OF PITCH, SPECTRUM, SPACE, AND RHYTHM

Roads: Could you describe your theory of spectral organization in more detail?

Dashow: The basic concept is using pitches to generate these inharmonic spectra, in the same sense that one would conceive of pitches as being accompanied by chords. Each spectrum is considered to be a chord and is treated as a chordal structure. With certain of these chords I can prolong these pitches, because they are literally (physically) present to a varying degree in these chord spectra. At the same time I give the pitches a range of harmonizations. This would be comparable to a melody tone that you hear in any voice and a piano composition where the voice holds a particular pitch and the pitch is harmonized by a series of chords that form different relationships to the held pitch.

Roads: You have discussed your compositions at considerable length in terms of pitch organization. But what strikes me about many of your compositions is at once the sense of rhythmic vitality and the exploration of musical

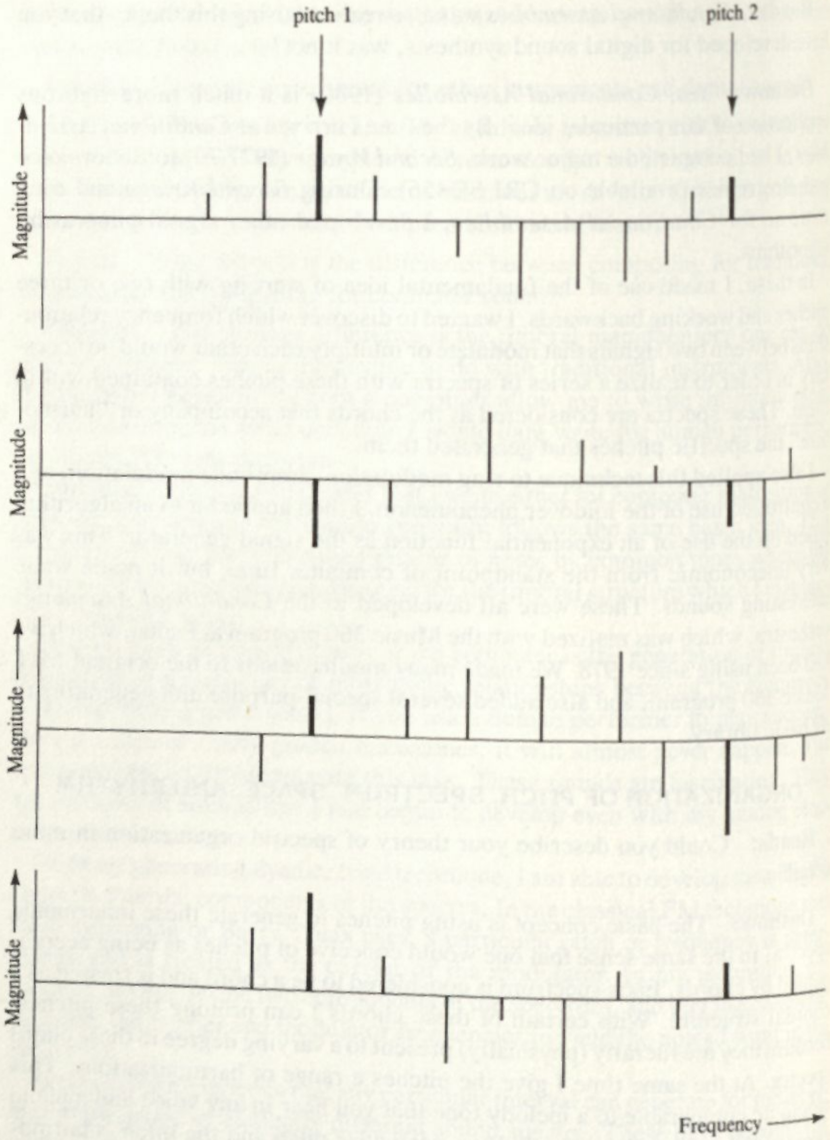


Fig. 3.1. Different hypothetical chord-spectra based on a generating dyad of pitch 1 and pitch 2. At successive times, the ratio of the amplitudes of the various accompanying frequencies with the dyad shifts, and different frequencies accompany the dyad.

space. Do you have a system for these parameters or are they organized intuitively?

Dashow: I apply a combination of intuitive and systematic means. Interpreting Petrassi's suggestion, I have found that creating a sense of space with electronic means can be a very dramatic event, no matter what kind of musical material is being "spatialized."

This spatial idea is combined with a rhythmic aspect and is used to delimit certain compositional ideas. In particular, the background notion of all of my pieces has to do with relationships between note groups. When I go from one kind of note group to another, an obvious way of suggesting I have changed the note group is by changing the rhythmic organization.

Roads: So then you see the rhythmic structure as always playing a kind of subordinate role in the service of pitch structure?

Dashow: Yes, that's exactly right. I try to make the pitch structure clear by applying my theory of spectrum derivation. The pitch structure on one level provides the background for the generation of the surface qualities—that is, the timbre, the colors, the harmonization, the chords that I use to surround the chosen pitches. In order to help clarify the pitch and chord-spectra relationships, I apply combinations of regular pulses and irregular pulses and various kinds of contrapuntal rhythmic mixtures.

The spatial notion is sometimes applied by placing some sounds further away and other sounds closer. When I am working with various note groups or chord spectra, I usually want you to hear a particular combination. I can separate the various groups with localization techniques. If something is closer, it is obviously one group, and something that is farther away is obviously another group. In this way it is possible to avoid the fusion of complex sounds into one sound when there are really several kinds of sounds going on simultaneously.

CLICHÉS OF COMPUTER MUSIC

Roads: What are the worst clichés in computer music?

Dashow: Certainly a common cliché and one easily abused is the "waves of sounds" idea where we are treated to a catalog of FM sounds. The waves come at us in long slow gestures, each one following the other without any sense of organization or direction.

The other abuse of the computer is the "one-idea composition," where the composer has discovered one sound and gives it to us for a half-hour. This is deadly!

Roads: If you were to cite one criticism that you thought was valid of your own compositional work, what would that be? In other words, what are you working to improve in your music?

Dashow: One of the real questions of my systems is: How does the listener decide which parts of the spectrum are the central pitches and which are the chordal accompaniment? How do you know which parts of the sound count structurally and which others are secondary or supportive?

Roads: Do you expect some further development of your theory of spectral organization will rescue you from this problem?

Dashow: I don't know. I am wondering if one could not abstract scalar properties from the frequency relationships of these spectra, such that each spectrum could represent pitch combinations from an inharmonically derived scale.

This concept has all kinds of implications. It suggests parallels to the way the church modes were organized. Here each inharmonic ratio generates a highly distorted nonequal temperament.

Then I can control the computer's tuning with algorithms that maintain the desired interval ratios as I go along. It's a simple matter of data manipulation. Perhaps I can treat these chord spectra as being the basis for hierarchically related scale systems. When I have a free moment, I intend to play around with this idea of abstracting entire scales from the several kinds of spectra that I like to work with and that I find to be extremely rich when used in combination. At any rate, I see the problem as a musical challenge. I think the system offers opportunities to resolve its problems, and the successful solution will be simply a convincing musical composition.

I am using with increasing regularity certain kinds of sounds which my generating dyad technique produces. For each piece, I manage to find ways of combining these sounds both vertically (in chords) and horizontally (in sequences and phrases). I want to widen the notion of working backward from the timbre to working backward from a larger musical context, and find the elements that can regenerate the context in a variety of ways. I want to decompose sounds to see if I can develop musical implications from the individual components "inside" the sounds. I see this as a sort of unfolding and recomposition of the chord spectrum, like looking at a lovely and complex flower through a kaleidoscope.

COMPUTER-ASSISTED COMPOSITION

Roads: What do you think of attempts to automate or simulate compositional process?

Dashow: I don't do it. I think composers who spend time automating their compositional procedures probably can spend their time better actually composing. However, I am willing to be convinced by the music of someone who has successfully automated his procedures. After all, it's the music that counts, not how it was produced.

Roads: What do you think of stochastic music?

Dashow: The label *stochastic*, like labels for all kinds of other procedures, does not really tell us what we are hearing. As a result, just like many other techniques that have been developed in this century, it is merely something for the composer to hang his hat on, something to get him going, something equivalent to his cup of coffee in the morning and his glass of wine at night or whatever he needs to keep going.

If you need a name like stochastic music to get your compositional ideas in order, fine, do it. I don't think any of these labels or any of these techniques really tell us what we are hearing. What a composer says about his music very often has nothing to do with what the music really tells us.

Roads: Does this apply to your own work as well?

Dashow: I am aware that maybe the theory I am pursuing might not have anything to do with what one eventually hears. I think I can hear in a convincing fashion the theoretical ideas that I propose. Above all, my idea is concerned with being able to hear the sounds in their composed relationships, not with an abstract procedure where the pitches don't count.

On the other hand, when my preconceived notions don't seem to work out all right, I will rely completely on my ears, as opposed to trying to force my music into an unproductive ideological bag. It's a very difficult question, because after you have worked with a musical idea for a long enough time you begin to hear in those terms. My music is motivated by trying to get you to hear what I think are interesting intervallic and timbral relationships. I want you to feel it, not just think about it. Procedures for procedures' sake are a delusion or just silly.

COMPUTER MUSIC AND OPERA

Roads: I understand you are now engaged in the composition of a large-scale work, an opera with computer sound and several singers. Could you tell us about the origin of the piece and your current state of work with it?

Dashow: The opera is based on *The Little Prince* by Antoine de Saint-Exupéry, and is a project I have had in mind since 1968. It is written for seven live voices, multichannel digital sound, and will involve the theatrical use of laser projections and computer graphics. All of this is to be synchronized by a master digital tape or by a computer in residence.

It is my first attempt at imagining a mixed-media piece. Up to now I have not been very favorable to the idea of trying to mix the extreme sophistication of the visuals with the extreme sophistication that contemporary music is capable of providing. I now see that perhaps the computer is going to provide us with a solution for successfully balancing visuals and music in the theater. Somehow through the absolute precision of the computer there is a possibility of synchronization that will allow a true integration of the two kinds of art which has been impossible in the past.

Most opera is a theatrical and visual success and a dismal bore musically, or vice versa. Computer sound and imagery are unique media which have obviously not existed before, and I think the results will also be unique.

Roads: Tell us about the musical organization of your opera.

Dashow: *The Little Prince* carries out my chord-spectra ideas in a rigorous manner. I have again increased the capacity of my computer orchestra with new algorithms. When writing an opera I have discovered you need an enormous quantity of material. Hence, I have decided to expand my ideas about timbral manipulations to include as great a variety as possible. I am including even diatonic collections of frequencies as one of many possible timbral possibilities that we are capable of listening to in an organized way.

Roads: Does this return to the diatonic scale represent a departure from your earlier style?

Dashow: I am not returning to the diatonic scale by any means. Nor am I returning to functional tonality. I am merely considering the diatonic collection of frequencies as one of the many options that we have available, such as combinations of the total chromatic, combinations of inharmonic frequencies with harmonically related frequencies, and so on.

Certainly when you are considering a piece of operatic dimensions—and this piece will be close to three hours long—you write more theatrically. You have to conceive of the music in other terms than when you write a concentrated piece for concert performance. The first and most important aspect of a large-scale theater piece is variety. I believe I can achieve the necessary variety by expanding my "source timbres." This leads me to include the diatonic collection along with the other kinds of sounds I have developed up to this point.

Roads: Tell us about the story of the opera.

Dashow: The first act includes the Little Prince's voyage among the planets. He finds on each planet a different character—an exaggeration of some

aspect of the human personality. He meets a King, a Drunk, a Business Man, a Lamplighter, a Conceited Man, and a Geographer. The Prince has a little conversation with each one of them. Finally he asks the Geographer where he should go to learn more about life. The Geographer suggests that he go to the planet Earth. It seems like an interesting planet, with several hundred kings, several thousand business men, millions of drunks, and billions of conceited people.

The finale of the first act is the Little Prince's voyage from the Geographer's planet to the planet Earth. This is a laser spectacular in which the audience has the feeling that they are out in space, accompanying the Little Prince on his voyage through the cosmos until he arrives on Earth.

I am quite excited about the idea of creating space with sound, but also with the notion of creating space visually. I want to blend the two sensations of visual space and sonic space, to involve the audience physically with this space voyage. Paul Earls, who is at the Center for Advanced Visual Studies here at MIT, and I are exploring ways of realizing the laser project. Paul has been working with computer-controlled lasers for several years. Some of the images he succeeds in creating are exactly what I need.

Roads: How does the plot of the opera relate to the musical structure of the piece?

Dashow: I decided to make a set of variations out of this voyage insofar as each planet represents an aspect of a human personality. Each planet is separated by a ritornello, and even the ritornelli go through an analogous kind of variation. The variations are in both pitch structure and in timbral realizations. Beyond this, each of the characters on the planets is portrayed with a five-note group. These five-note groups have slight differences which allow me to develop them into separate vocabularies. In this way, I manage to characterize each of their personalities in sound.

Several of these groups are so rich in musical possibilities that I can foresee separate concert pieces based on them. For example, *Mnemonics* for violin and computer sounds is based on the Conceited Man group and how it interacts with the Little Prince's group. I hope this isn't a foreshadowing of my relationships with violinists!

IN WINTER SHINE

Roads: I understand you've just completed a composition called *In Winter Shine* realized at MIT. Could you tell us your goals with this new composition?

Dashow: The first goal was to compose a piece of more chamber music proportions rather than one of symphonic proportions, as I have done for the past few works. I wanted to think in more compact and concentrated terms, to see if my pitch prolongation by means of chord spectra could be accomplished on a smaller time scale and in a more succinct fashion. And practically speaking, it has become clear that shorter pieces are necessary for meeting the requirements of certain concert situations.

Roads: Does this piece represent a new direction for you compositionally, or is it one in a series of works that explore the same ideas?

Dashow: It's a modification of some earlier ideas for underlying structure. I am using note groups in a different fashion than I have in the past, but I am continuing to develop the idea of inharmonic chordal accompaniments to these particular pitches. I see enough material in this approach to last me for perhaps the rest of my life.

APPENDIX: COMPOSITIONS BY JAMES DASHOW

Songs of Despair (1968–69). Soprano and eleven instruments.

Timespace Extensions (1969). Piano, flute, and two percussion.

Duo (1970). Violin and piano.

Astrazioni Pomeridiane (1970–71). Orchestra.

Ashbery Setting (1971–72). Soprano, piano, and flute.

Burst (1971). Soprano and electronic sound.

Maximus (1972–73). Voice and chamber ensemble with piano and percussion.

Mappings (1974). Cello and electronic sound.

Some Dream Songs (1974–75). Soprano, piano, and violin.

At Delphi (1975). Voice and electronic sound.

Punta di Vista, No. 1, Forte Belvedere (1975–76). Piano.

Whispers Out of Time (1976). Electronic sound. Winner of First Prize at the Bourges International Festival, 1979.

Effetti Collaterali (1976). Clarinet and computer-generated sound.

Punta di Vista, No. 2, Montiano (1977). Piano.

A Way of Staying (1977). Soprano and computer-generated sound.

Second Voyage (1977–79). Tenor and computer-generated sound. Commissioned by the National Endowment for the Arts.

Partial Distances (1979). Electronic sound.

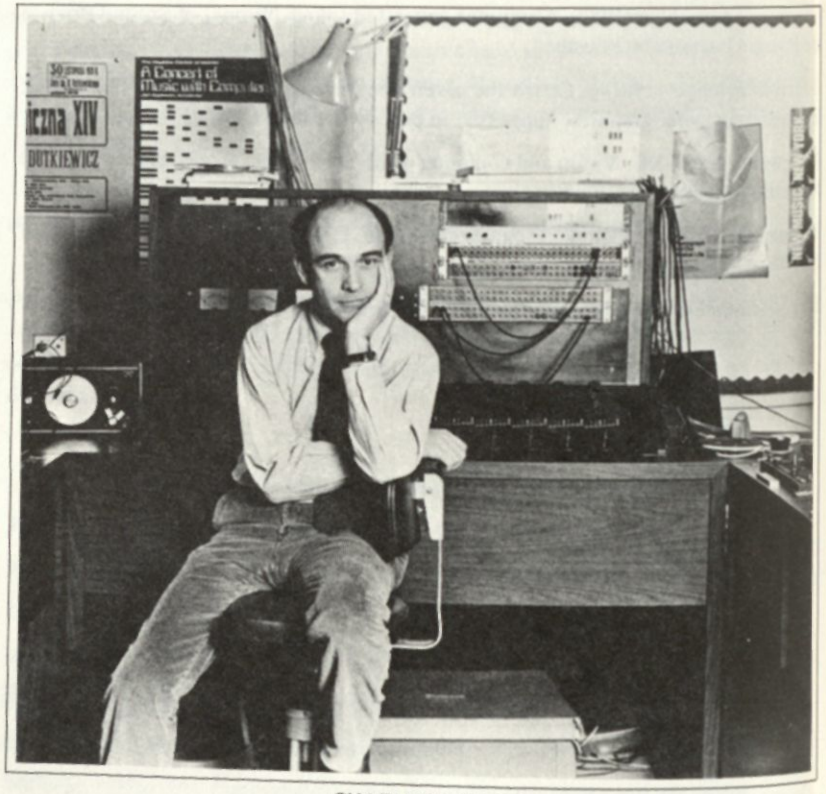
Conditional Assemblies (1980). Computer-generated sound. Winner of Second Prize at the Bourges International Festival.

Il Piccolo Principe (1981–). Opera for seven vocalists, computer-generated sound, lasers, and computer graphics. Supported in part by the Rockefeller Foundation.

Mnemonics I (1981–84). Violin and computer-generated sound. Commissioned by the National Endowment for the Arts.

In Winter Shine (1983). Computer-generated sound. Commissioned by the Council for the Arts at MIT.

Sequence Symbols (1984). Computer-generated sound.



CHARLES DODGE

4

IN CELEBRATION: THE COMPOSITION AND ITS REALIZATION IN SYNTHETIC SPEECH

Charles Dodge

In Celebration, an electronic music realization of the poem by Mark Strand (1973), was composed during the first half of 1975 and realized at the Columbia University Center of Computing Activities and the Nevis Laboratories (Dodge 1975).

In 1971, I began to investigate the compositional uses of computer-synthesized speech. The method used is known as *speech synthesis by analysis*, which bases the synthetic speech on a recorded voice and allows for modification of aspects of the voice analysis before synthesis. (See Cann 1979, 1980.) Using this digital technique, it is possible to extend the treatment of the recorded voice beyond the scope of traditional tape music. This new medium attracts me because it brings together the potential of computer music for variation over a wide range of timbre, pitch, and rhythm, with *musique concrète's* emphasis on the unique acoustical features of language and speaker.

In this piece, the poem simultaneously serves as the textual and acoustical source for its own setting. All of the sounds in the composition are made with computer synthesis derived from a single reading of the poem. Thus the piece expresses the poem with a directness that is not often found in traditional musical settings—that is, singer(s) with accompaniment. The situation of *In Celebration* is more closely related to a male actor's reading of the poem (albeit an actor who can extend his voice over five octaves, conjure up multiple copies of his voice at will, control the pitch and rhythm of his voice to an extraordinarily fine degree, and perform other vocal and musical tricks). The use of a single human voice as the foundation for all the sounds in the piece lends unity to the composition.

THE COMPOSITION

The musical realization of *In Celebration* reflects the two-part structure of the poem (see fig. 4.1), dividing after the second occurrence of "You sit in a chair." The two parts of the poem describe different psychological orientations of the subject. In the first part, the subject appears devoid of emotion or the ability to act—as though dead. The language conveys an attitude of hopelessness, despair, and stasis. The second part, while continuing the tone of the first,

underlines a heightened self-awareness. Awaiting death becomes a celebration, and a sense of resolution pervades the latter half of the poem.

The music portrays this change of emphasis. The first part treats the text with various types of articulation, including spoken, whispered, pitched, and glissandoing phrases, and with textures from solo to choral. These various forms of articulation follow in rapid succession, and there is a prevalence of textures that include more than one type of articulation, reflecting the indecisiveness of "You." In contrast, at almost any given moment in the second part, the type of articulation is uniform. Uncombined, the different sorts of articulation convey greater resolve.

IN CELEBRATION
by Mark Strand

You sit in a chair, touched by nothing, feeling
the old self become the older self, imagining
only the patience of water, the boredom of stone.
You think that silence is the extra page,
you think that nothing is good or bad, not even
the darkness that fills the house while you sit watching
it happen. You've seen it happen before. Your friends
move past the window, their faces soiled with regret.
You want to wave but cannot raise your hand.
You sit in a chair. You turn to the nightshade spreading
a poisonous net around the house. You taste
the honey of absence. It is the same wherever
you are, the same if the voice rots before
the body, or the body rots before the voice.
You know that desire leads only to sorrow, that sorrow
leads to achievement which leads to emptiness.
You know that this is different, that this
is the celebration, the only celebration,
that by giving yourself over to nothing,
you shall be healed. You know there is joy in feeling
your lungs prepare themselves for an ashen future,
so you wait, you stare and you wait, and the dust settles
and the miraculous hours of childhood wander in darkness.

Fig. 4.1. Mark Strand, "In Celebration" from *The Story of Our Lives*. Copyright © 1973 Mark Strand. Reprinted with the permission of Atheneum Publishers.

Part I

"You" is the primary recurring word in the poem, and it becomes the central unifying theme of the first half of the musical composition. Because it consists of a diphthong, it entails, by definition, a transition from one vocal articulation position to another. This "timbre change" is used to develop most of the abstract, "nonvocal" passages in the piece. The word is associated with single pitches and with simultaneous and successive collections of pitches (chords and melismas). It is also accorded treatment by glissando, a type of articulation that here represents a halfway point between spoken and pitched speech. The proliferation of different settings of the word "You" represents an association of maximum pitch activity in the piece with a minimum of text. (See measures 1–4 and 15–24 of fig. 4.2.)

The phrase "in a chair" is a pitched line that provides a definitive sound at the beginning and end of the first part of the piece. "In a" is articulated heterophonically in two voices, one pitched and the other as a glissando (measure 7). "Chair" (measure 8) is elaborated by octave duplication and contrast of duration in its five voices. For the second occurrence of "in a chair" (measures 56–57), the pitches for "in a" are inverted and the longer notes of "chair" (again, on B-flat) are placed below the shorter ones in about the same time proportions. This musical statement is repeated near the end of the composition by means of the phonetically similar phrase "there is joy," which also occurs twice in a similar relationship.

An example of a chorus mixing types of articulation can be heard in the setting of "touched by nothing" (measure 9), which combines pitched, spoken, and glissando contours in staggered rhythmic patterns. The setting of "feeling the old self become the older self" (measure 10) employs a more elaborate heterophony than the setting of "in a." Here there are two rhythmically contrasted voices, one a glissando and one pitched over the same interval span. The pitched voice proceeds more and more slowly to the end of the phrase, whereas the glissando proceeds at a constant speed.

"Imagining only" (measure 11) introduces a texture used extensively later in the work: three equal-tempered voices in chords. The two phrases "The patience of water, the boredom of stone" (measure 12) are treated in a straightforward text painting by use of glissandi. Interruption, a device found throughout the work, appears here in the repetition of "imagining" (measure 13) interjected between the successive and simultaneous glissandi.

"Silence" (measure 26) is treated in a chorus reminiscent of the one used for "chair." Within each chorus, all the voices have the same quality of pitch, but "silence" is realized with noise, whereas "chair" is sounded in octaves. The

Fig. 4.2. Score of *In Celebration*, a musical composition by Charles Dodge (available on CRI record number SD348). Copyright © 1975 Charles Dodge. Cape North Music. Used with permission.

Fig. 4.2. *Continued*

Fig. 4.2. *Continued*

Fig. 4.2. *Continued*

Fig. 4.2. *Continued*

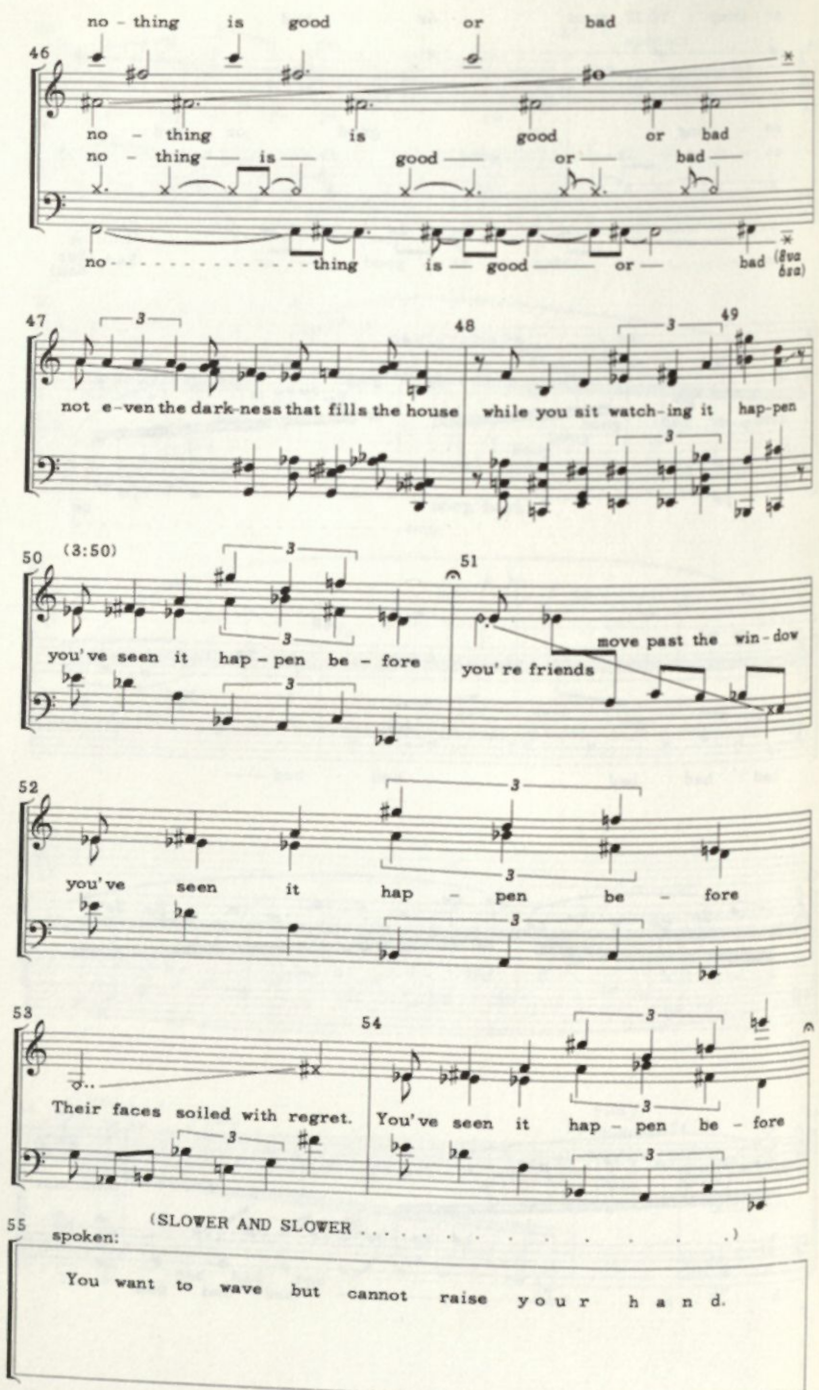


Fig. 4.2. *Continued*

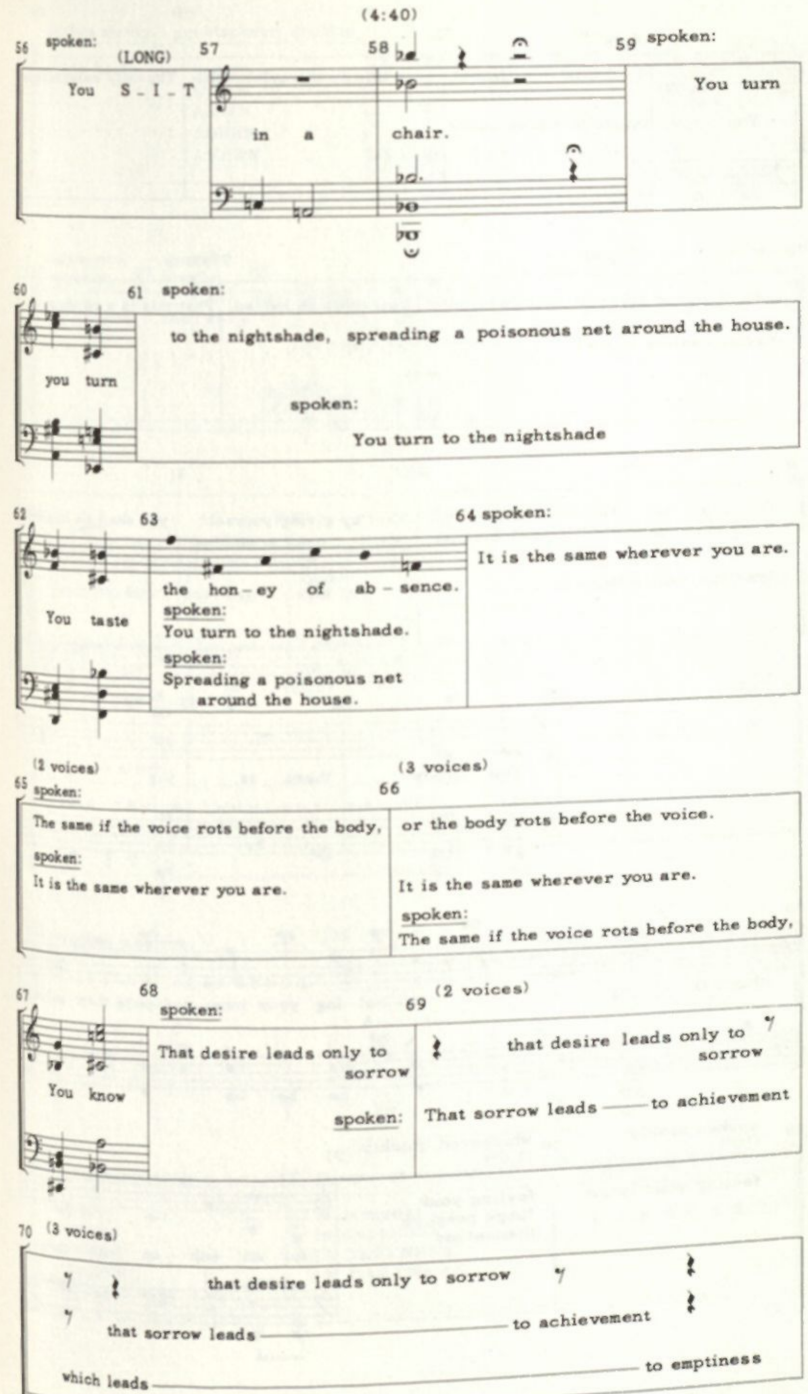


Fig. 4.2. *Continued*

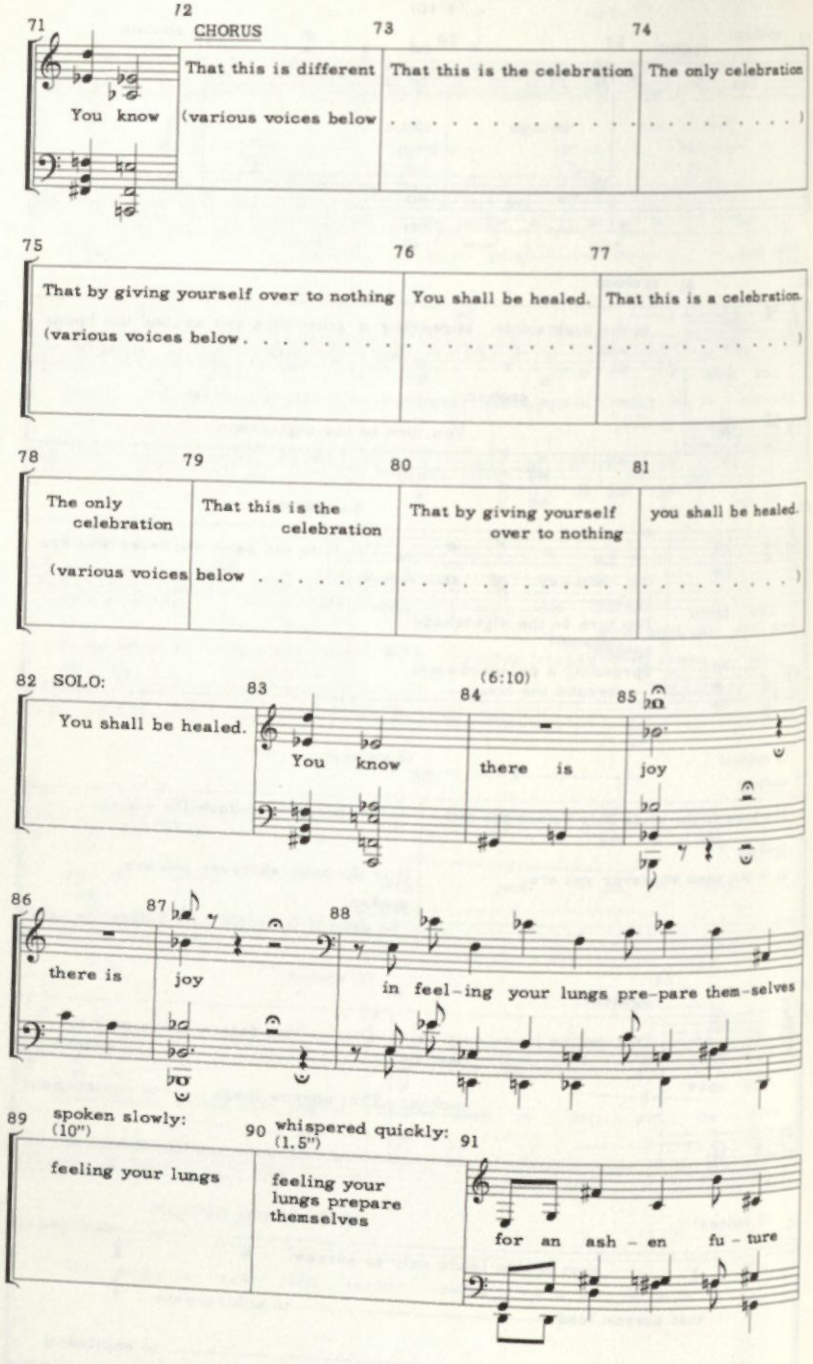


Fig. 4.2. *Continued*

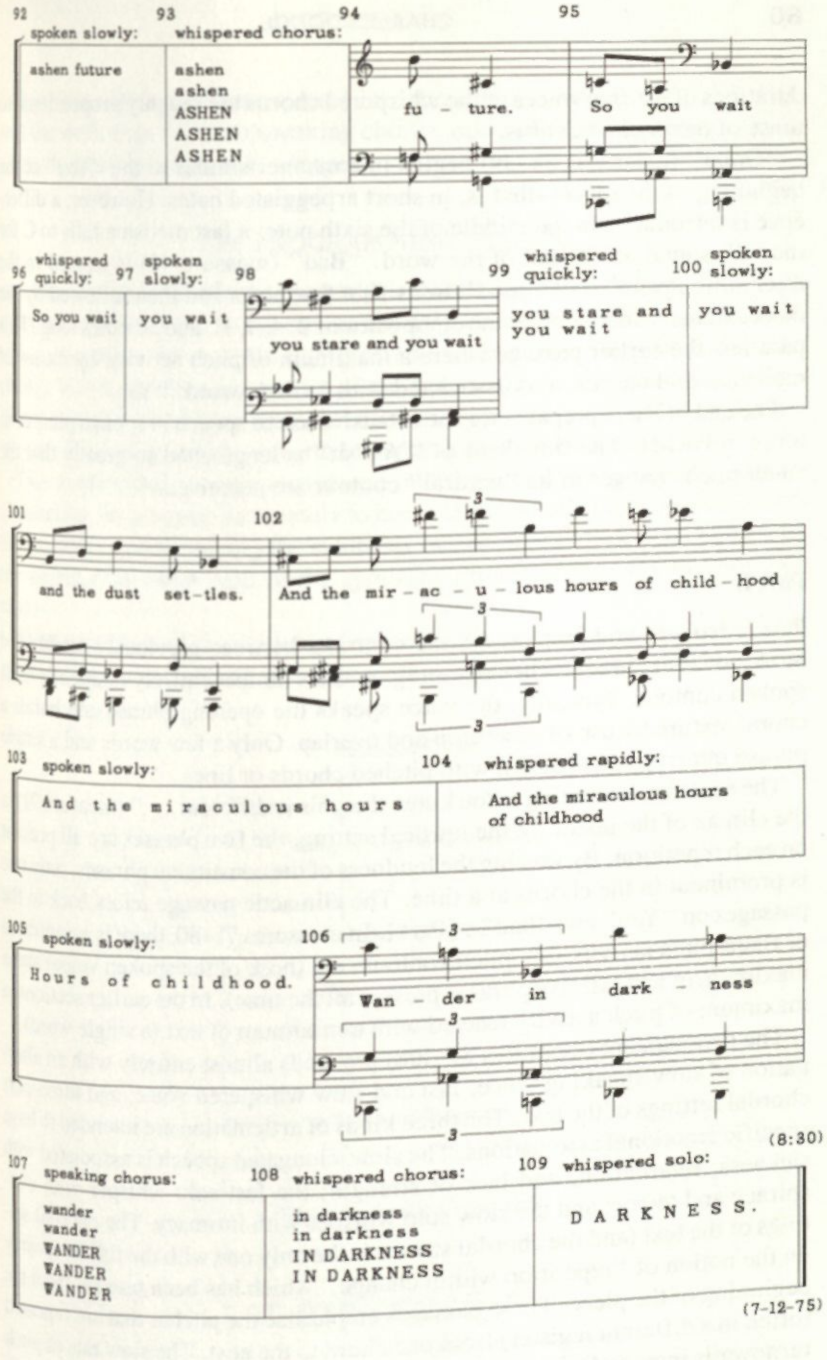


Fig. 4.2. *Continued*

durations of the five voices of the whispered chorus are roughly proportional to those of the pitched chorus.

"Good" (measures 38–39) begins in a manner similar to the "You" at the beginning of the piece—that is, in short arpeggiated notes. However, a difference is introduced in the middle of the sixth note: a fast melisma falls to C for the "d" sound at the end of the word. "Bad" (measure 40) is treated in the intervallic inversion of "good" in its first few notes and then followed by the most extensive melismas of the composition: 2, 4, 7, 11, and 52 notes long. This parallels the earlier passage where a maximum of pitch activity, by means of melismas and chords, was associated with a single word, "You."

The end of Part I prepares for the extensive use of speech in spoken pitch contours in Part II. The time base of "You sit" is lengthened so greatly that the small pitch changes of its "natural" contour are perceived.

Part II

Part II isolates and emphasizes the contrasts between previously established modes of articulation. The beginning consists almost entirely of speech in its spoken contour. Typically, the voice speaks the opening phrase and builds a choral texture by use of repetition and overlap. Only a few words and a single phrase interrupt this pattern with pitched chords or lines.

The sentence beginning "You know that this is different . . ." (measure 71) is the climax of the poem. In the musical setting, the five phrases are all present on each repetition. By varying the loudness of the constituent phrases, only one is prominent in the chorus at a time. The climactic passage refers back to the passages on "You" and "bad" of Part I. In measures 71–80, there is a minimum of fixed-pitch activity (the pitch contours are those of the spoken voice) and a maximum of text (all five phrases present all the time). In the earlier sections, a maximum of pitch activity reacted with a minimum of text (a single word).

The denouement (measures 87–108) proceeds almost entirely with an alternation of slow speaking voice, fast and slow whispered voice, and three-part chordal settings of the text. The three kinds of articulation are intended to have specific emotional associations. The slow, elongated speech is associated with sickness, retardation, and lack of strength; the fast solo whisper with conspiracy and terror; and the slow solo whisper with intimacy. The chordal settings of the text (and the chordal stream is the only one with the full text) carry on the notion of "repetition within change," which has been present from the beginning of the piece. These passages emphasize the pitches that are repeated (often in a different register) from one chord to the next. The slow rate of pitch turnover is intended to convey a feeling of very slow evolution and conclusion.

A whispered chorus is interjected into the alternation of the three streams, and the work ends with one speaking chorus, one more whispered chorus, and a greatly time-distorted solo whisper.

THE PROGRAMMING SYSTEM

An overview of the synthesis-by-analysis system is presented in figure 4.3. In this process, the speech is digitized at 15 000 samples per second, and the analysis program transforms the digitized speech into a set of parameters that change with time. From the output of the analysis program, the synthesis program recreates the digital representation of the voice at 15 000 samples per second. The voice is then converted to analog form and recorded.

Two major techniques are introduced into the system for musical purposes: (1) altering the analysis parameters to cause the synthetic voice to differ from the original, and (2) mixing the synthesis output (stored on digital tape) with the output from other runs of the synthesis program (also stored on digital tape).

The cornerstone of the method of speech synthesis is the analysis program, which reduces the digitized speech to a set of parameters through two separate

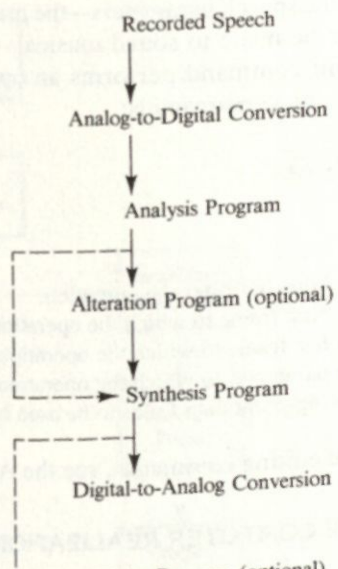


Fig. 4.3. An overview of the synthesis-by-analysis system used for *In Celebration*

processes: (1) extracting the coefficients of an all-pole filter by the *linear predictive coding* (*LPC*) method (Markel and Gray 1976), and (2) *tracking the spoken pitch contour*.

The analysis progam reduces the speech data from 15 000 samples per second to a set of 120 *frames* per second. Each frame (analogous to a motion picture frame) is described by 24 parameters: the low-frequency amplitude, the high-frequency amplitude, a ratio of the two amplitudes that is used for determining whether the frame is voiced or unvoiced, the pitch, the 19 coefficients for the all-pole filter, and the duration of the frame.

The synthesis stage of the program is illustrated in figure 4.4. The first parameter required is the frame length, which sets the number of output samples to be generated from a single set of parameters. The next parameter determines whether the frame is to be voiced or unvoiced. For a voiced frame, the synthesizer uses the pitch parameter as input to a pulse generator that simulates the glottal wave. For unvoiced, the synthesizer uses a noise generator to simulate turbulence in the vocal tract. The output of the appropriate generator, multiplied by the high-frequency amplitude parameter, serves as input to the all-pole filter, which simulates the resonances of the vocal tract. The output of the filter is the digital representation of the synthetic-speech waveform, which is converted to analog form on audio tape.

The option of altering the speech parameters—the principal means by which the synthetic speech can be made to sound musical—is implemented in an editing program. An edit command performs an operation on a range of frames. The general form of a command is:

```
C, I1, I2, j, E1, E2, E3
```

where

- `C` is a control character that indicates the operation.
- `I1` is the number of the first frame to which the operation applies.
- `I2` is the number of the last frame to which the operation applies
- `j` is the number of the parameter to which the operation applies.
- `E1`, `E2`, and `E3` are floating-point data fields to be used by the operators.

(For a list of the available editing commands, see the Appendix.)

THE COMPUTER REALIZATION

The composer uses the editing program to create musical voices out of the frames of analyzed speech. The first step in shaping the frames into musical passages is to search through the printed record of the analysis for word and syllable boundaries of the spoken text. Figure 4.5 shows the printout for the first

55 frames of the words "You sit" as spoken at the beginning of the poem. The printout includes the high-frequency amplitude (RMS2), the low-frequency amplitude (RMS1), their ratio (ERRN—which is used in determining whether the frame is voiced or unvoiced), and the fundamental frequency of the voice (PITCH). From these parameters it is possible to scan the speech for word and syllable boundaries with considerable accuracy. Once the syllable boundaries

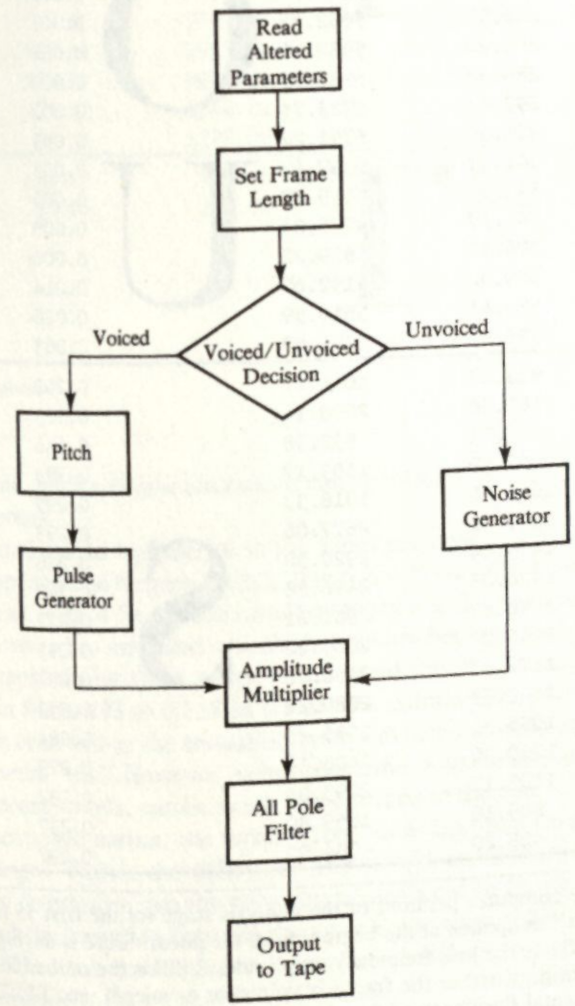


Fig. 4.4. Linear predictive coding (LPC) synthesis stage

YOU

FRAME	RMS2	RMS1	ERRN	PITCH
175	58.70	848.46	0.005	81.95
176	62.51	2241.17	0.001	81.95
177	69.53	2021.38	0.001	164.81
178	134.07	2271.37	0.003	164.81
179	118.87	2564.07	0.002	129.30
180	146.32	2855.66	0.002	129.30
181	211.07	3262.10	0.004	138.87
182	186.01	3864.62	0.002	138.87
183	205.50	4512.10	0.002	142.84
184	247.11	5124.51	0.002	142.84
185	215.83	5682.63	0.001	147.04
186	282.20	5987.16	0.002	147.04
187	284.92	5983.75	0.002	150.00
188	293.65	5783.75	0.002	150.00
189	328.96	5751.27	0.003	141.50
190	341.53	5537.47	0.003	141.50
191	341.84	5159.77	0.005	141.50
192	341.17	4775.06	0.005	141.50
193	374.31	4539.22	0.006	140.17
194	509.02	4192.67	0.014	140.17
195	583.41	3616.59	0.026	140.17
196	578.87	2291.59	0.063	140.17

S

FRAME	RMS2	RMS1	ERRN	PITCH
197	813.27	1618.21	0.252	937.50
198	1189.36	2090.14	0.323	937.50
199	553.71	838.38	0.436	937.50
200	742.59	1183.17	0.393	937.50
201	1041.95	1918.33	0.295	123.95
202	1449.16	2677.06	0.293	123.95
203	1454.84	2920.50	0.248	937.50
204	1430.03	2496.88	0.348	937.50
205	1570.88	2981.21	0.277	142.84
206	1443.27	2665.22	0.293	142.84
207	1172.67	2150.50	0.297	150.00
208	1200.73	2080.20	0.333	150.00
209	1095.51	2055.25	0.284	116.26
210	1260.36	2408.14	0.273	116.26
211	1105.17	2293.05	0.232	937.50
212	809.10	1659.80	0.237	937.50
213	428.20	784.93	0.297	250.00

Fig. 4.5. The computer printout of the analysis stage for the first 55 frames of the words "You sit" as spoken at the beginning of the poem. RMS2 is the high-frequency amplitude, RMS1 is the low-frequency amplitude, ERRN is the ratio of RMS2 to RMS1 (used to determine whether the frame is voiced or unvoiced), and PITCH is the estimated fundamental frequency of the voice.

I

FRAME	RMS2	RMS1	ERRN	PITCH
214	419.45	3886.15	0.011	250.00
215	925.86	6366.20	0.021	208.32
216	746.28	8046.81	0.008	208.32
217	829.82	8277.42	0.010	192.29
218	754.64	8049.50	0.008	192.29
219	771.84	8001.70	0.009	197.35
220	726.81	7955.17	0.008	202.69
221	807.63	7835.20	0.010	202.69
222	874.27	7732.59	0.012	205.42
223	776.87	7491.86	0.010	205.42
224	684.64	7317.04	0.008	205.42
225	560.87	6297.36	0.007	102.03
226	175.63	1842.81	0.009	102.03
227	46.35	1329.09	0.001	197.85

T

FRAME	RMS2	RMS1	ERRN	PITCH
228	38.25	793.00	0.002	197.85
229	39.26	316.92	0.032	202.69

Fig. 4.5. Continued

are determined, pitch and time alterations can be introduced into the speech for musical purposes.

The principal cue to boundaries is the ERRN parameter, although the RMS values and fundamental frequency (PITCH in fig. 4.5) can also be useful. In the case of "You sit" (fig. 4.5), the choice of boundary is a clear one because of the change from voiced to unvoiced speech that occurs between the words. ERRN exhibits the rapid change from voiced to unvoiced speech, increasing in value from 0.006 in frame 193 to 0.252 in frame 197. Similar dramatic changes of ERRN may be observed in the transition from unvoiced to voiced frames and back in the word "sit." However, some phoneme boundaries, such as those between adjacent vowels, can be much less obvious to the eye.

As was mentioned earlier, the word "You" is a diphthong that exhibits a "timbre change." Notice the difficulty one would encounter, however, in attempting to determine from the parameters displayed in figure 4.5 a single point at which the transition between the vowels had been achieved.

"You" is edited and synthesized in a number of different ways for *In Celebration.* In measure 1, the six short "You" sounds were created by editing the analysis with the following commands:

```
T  175  196    0.2          P  175  196   9.04
Z  175  196  219.0          P  197  218   8.07
Z  175  196  241.0          P  219  240   7.08
Z  175  196  263.0          P  241  262   6.11
Z  175  196  285.0          P  263  284   7.10
                            P  285  306   8.09
```

The T (TIME) command sets the duration of each "You" to 0.2 sec. The four Z (MOVE FRAMES) commands copy all the parameters of the 22 frames of "You" four times. The P (PITCH) commands set the pitch for each note.

The chord in measure 1 was created by separately synthesizing the five voices, each with the same duration (c. 1.5 sec) and digitally mixing the five parts of the chord.

Measure 2 was made in an identical fashion using different values for some of the E1's.

The glissando in measure 3 was achieved with two editing commands:

```
T  175  196  1.0
I  175  196  4    204.    370.
```

The T command sets the duration and the I (INTERPOLATE) command causes the frequency in hertz to change linearly from 204 Hz in frame 175 to 370 Hz in frame 196.

The melisma in measure 4 was created by these commands:

```
T  175  196  2.0
P  175       9.03
P  176       9.06
P  177       9.05
   .
   .
   .
P  196       6.08
```

The T command sets the frame rate to 11 per second and the P command is applied once to each of the 22 frames. The resulting "electronic melisma" embodies the same change of timbre in two seconds as the diphthong of the original "You."

The "sit" of measure 5 was created in a similar fashion, but with multiple lines for overlapping some of the notes. The separately created lines were

mixed digitally. The commands for the most active of the constituent lines follow:

```
T  197  227  3.00
P  214       9.07
P  215       9.06
P  216       9.00
   .
   .
   .
P  227       6.08
```

Notice that the T command applies only to frames 197–227, and does not include the frames occupied by the final "t." The last phoneme is synthesized at the original frame rate to avoid interfering with the clarity of the plosive, which could be adversely affected by time distortion.

It can be seen from the foregoing how the editing facility of the analysis/synthesis system enables one to make music out of a recorded voice. The articulation of the speaker and the features imparted by the analysis system itself play a significant role in shaping compositional material. For example, the "You" was analyzed into 22 frames, and this fact accounts for the number of notes in the melisma of measure 4. There are numerous other examples in the work where the number of frames representing a particular phoneme will account for its musical treatment (e.g., measures 5, 15, 16, 18, 20–21, 29–33, 39, 42–43).

The two voices of measure 6 treat "in a" heterophonically. The voice on the lower staff in the score was made by the T and P commands. The voice on the upper staff, although occupying the same time and interval spans, was created by the D (DILATE) command for its timing and the G (GLISSANDO) command for its pitch.

The "chair" chorus was created by digitally mixing its separately computed five voices. The T command was applied equally to the voiced and unvoiced phonemes so that the noise of "ch" on the very long, high B-flat would form a background for the shorter versions of "chair" in the chorus.

The chorus on "touched by nothing" (measure 9) combines all the types of pitched articulation used in the work. No two voices change syllables at the same time and all have contrasting pitch contours: the soprano alternates between two pitches, the alto consists of a rising glissando, the tenor plays the greatly time-distorted natural pitch contour of the voice, and the bass plays a descending glissando.

Measure 10 was realized by first synthesizing the fixed-pitch voice with P commands for its melody and with the I command to cause the frame rate of the voice to decrease gradually. The glissando voice was made to occupy the same time span as the pitched voice by means of the T command. The G command was used to cause the voice to begin and end with the pitches of the first voice and to glissando between them.

The W (TRILL) command was used to obtain the alternation of the G's and A-flats at the end of measure 43. The D command was used to gradually slow down the frame rate.

The chorus that constitutes the climax of the composition (measures 71–80) was created in the following way: the five phrases of the text were synthesized using the pitch contour of the recorded voice, and all were altered by means of the T command to occupy the same duration, 2 sec. Before each mix, one of the five voices was increased in amplitude by means of the B (BOOST) command so that it was 6 dB greater than each of the others. This has the effect of pushing the other voices into the background for the duration of the phrase, causing the "boosted" voice to carry the meaning of the text for the whole phrase.

In the last measure of the piece, the solo voice is altered by changing all the frames to unvoiced, creating a whisperlike effect. This is accomplished by setting the ERRN parameter to 0.1 by means of the C (CHANGE) command. The T command rendered the duration of the word "darkness" from its original 0.5 sec to 15 sec.

CONCLUSION

This paper has indicated some of the technical and musical problems addressed in the creation of *In Celebration*. Although it reinforces and extends the meaning and emotional impact of the poem, the composition is uniquely designed to be realized with synthetic voices. The programming system implements sophisticated algorithms for resonance and pitch analysis and provides a means for altering speech and making it sound musical. The computer realization uses the altering and mixing features of the programming system to record a virtuosic performance.

ACKNOWLEDGMENTS

The LPC program was written by Kenneth Stieglitz of the Department of Electrical Engineering of Princeton University. The pitch-tracking algorithm was programmed by Howard Eskin of the Columbia University Department of Electrical Engineering and Computer Science. Richard Garland, Thomas A. Jerse, Darius Clynes, and Charles Dodge contributed to the editing program.

REFERENCES

Cann, R. 1979, 1980. An analysis/synthesis tutorial. *Computer Music Journal* 3(3):6–11, 3(4):9–13, 4(1):36–42. Revised and updated version in C. Roads and J. Strawn, eds. 1985. *Foundations of computer music*. Cambridge: MIT Press.

Dodge, C. 1975. In celebration. On the record *Charles Dodge: Synthesized Voices*. New York: Composer's Recordings Incorporated. CRI SD348.

Markel, J., and A. Gray, Jr. 1976. *Linear prediction of speech*. New York: Springer.

Strand, M. 1973. In celebration. In *The Story of Our Lives*, 19. New York: Atheneum.

APPENDIX: EDITING COMMANDS USED IN THE REALIZATION OF *IN CELEBRATION*

The general form of a command is:

```
C, I1, I2, J, E1, E2, E3
```

where

- C is a control character that indicates the operation.
- I1 is the number of the first frame to which the operation applies.
- I2 is the number of the last frame to which the operation applies
- J is the number of the parameter to which the operation applies.
- E1, E2, and E3 are floating-point data fields to be used by the operators.

X—Examine

Prints value of parameter J for frames I1 through I2 on the terminal. E1, E2, and E3 have no effect. Example:

```
X 101 103 4
```

Prints out the value of parameter 4 for frames 101–103 as follows:

Parameter	Frame number	Value
4	101..	378.40012
4	102..	378.40012
4	103..	189.00000

C—Change

Changes value of parameter J for frames I1–I2 to the value given by E1. Example:

```
C 101 103 24 0.007
```

Changes the value of parameter 24 to 0.007 in frames 101–103.

I—**Interpolate**

Changes the value of parameter J in frame I1 to the value of E1, the value of parameter J in frame I2 to the value of E2, and the value of parameter J in all frames between I1 and I2 to a value that is linearly interpolated between E1 and E2. Example:

```
I 101 105 1   2.0     6.0
```

Changes the value of parameter 1 in the following frames as follows:

```
101  2.0
102  3.0
103  4.0
104  5.0
105  6.0
```

Z—**Move Frames**

Moves frames I1–I2 to replace frames starting with E1. If E2 and E3 are left blank, all parameters are moved. If E2 and E3 are specified, then for each frame only parameters E2–E3 are moved. Example:

```
Z 101 102 302.0
```

Replaces frame 302 with all parameters from frame 101. Replaces frame 303 with all parameters from frame 102.

B—**Boost**

Raises the amplitude of frames I1–I2 by the value of E1 in decibels. A negative value of E1 lowers the amplitude. Thus, the amplitude (parameter 1) is multiplied by $10^{(E1/20)}$. Example:

```
B 101 103    6.0
```

Boosts the amplitude of frames 101–103 by 6 dB—that is, the amplitude is doubled.

O—**Crescendo**

Boosts the amplitude of frame I1 by E1 dB and the amplitude of frame I2 by E2 dB. The amplitude of the frames between I1 and I2 are boosted by an amount linearly interpolated in decibels from E1 to E2. Note that since the interpolation is linear in decibels, the actual sequence of amplitude multipliers is exponential. Example:


```
O 101 105    0.0    4.0
```

Boosts the amplitude as follows:

```
101  0.0 dB
102  1.0 dB
103  2.0 dB
104  3.0 dB
105  4.0 dB
```

R—**Raise**

Raise the pitch of frames I1–I2 by the value of E1 in half steps. A negative E1 lowers the pitch. Example:

```
R 121 123    2.0
```

Raises the pitch of frames 121–123 by a whole step.

P—**Pitch**

Sets the pitch of frames I1–I2 to the value of E1 in *octave point pitch class* notation. In this notation the number to the left of the decimal point is the octave number and the number to the right, when multiplied by 100, is the half step within the octave. Middle C is defined to be 8.0 (261.7 Hz), so that A = 440 Hz is 8.09. An equal-tempered scale is used. Example:

```
P 115 117    7.03
```

Sets the pitch in frames 115–117 to E-flat below middle C.

G—**Glissando**

Raises the pitch of frame I1 by E1 half steps and frame I2 by E2 half steps. The pitches of the frames between I1 and I2 are raised by an amount linearly interpolated in pitch between E1 and E2. Since the interpolation is linear in pitch, the actual sequence of frequency multipliers is exponential. Example:

```
G 106 109    0.0    6.0
```

Raises the pitch as follows:

```
106  0.0
107  2.0
108  4.0
109  6.0
```

W—**Trill**

Raises the pitch of every other frame by E1 half steps beginning with frame I1 and ending with frame I2. When E2 is specified, every *k*th frame is raised (where $k = E2$). Example:

```
W 101 106   1.0
```

Raises the pitch of frames 101, 103, and 105 by one half step. The pitch of frames 102, 104, and 106 are not modified. Example:

```
W 101 120   2.0   5.0
```

Raises the pitch of frames 101, 106, 111, and 116 by two half steps. The other frames are not modified.

V—Voicut **Set**

Sets Voicut to the value of E1. Voicut is compared to ERRN (parameter 3) of a frame to determine if that frame is voiced or unvoiced. If parameter 3 is greater than Voicut, the frame is unvoiced. The default value of Voicut is 0.007. Since the command sets Voicut for an entire run of the synthesis program, I1 and I2 have no effect. Example:

```
V  .01
```

Sets Voicut equal to 0.01.

T—**Time**

Sets the time span occupied by frames I1–I2 to the duration of E1 in seconds. The time is divided equally among the frames. Example:

```
T 101 200   0.9
```

Sets the time span occupied by frames 101–200 to 0.9 sec by setting the duration of each frame to 0.009 sec.

E—**Expand**

Multiplies the duration of frames I1–I2 by the value of E1. If j is nonzero, only the duration of *voiced* frames is affected (that is, modify the frame only if parameter 3 of the frame is less than Voicut). Example:

```
E 111 136 0   0.5
```

Multiplies the duration of frames 111–136 by 0.5 (that is, runs the frames at twice the speed). Example:

```
E 111 136 1   1.5
```

Multiplies the duration of frames 111–136 by 1.5 only if the frame is voiced.

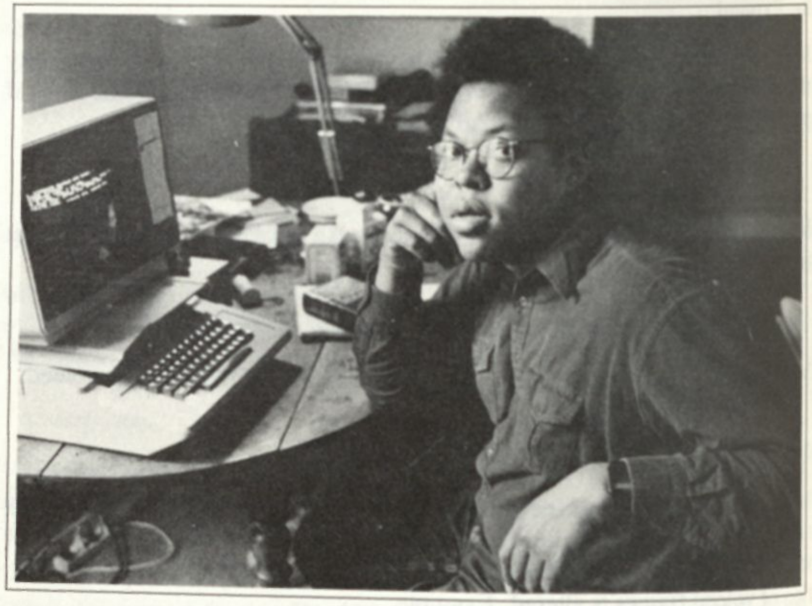
D—**Dilate**

Multiplies the duration of frame I1 by the value of E1. Multiplies the duration of frame I2 by E2. Multiplies the duration of frames between I1 and I2 by a value linearly interpolated between E1 and E2. If j is nonzero, only the duration of voiced frames is altered. Example:

```
D 101 105 0   1.0   2.0
```

Multiplies the duration as follows:

```
101  1.00
102  1.25
103  1.50
104  1.75
106  2.00
```



GEORGE LEWIS

5

IMPROVISATION WITH GEORGE LEWIS

Curtis Roads

George Lewis (born 1952 in Chicago) is a composer and trombonist based in Paris. He studied philosophy at Yale, and from 1980 to 1982 he was music program director at the Kitchen Center for Video, Music, Dance and Performance Art in New York City. This interview was conducted 9 November 1983 in New York, while Lewis was visiting to perform in a concert featuring the music of Earl Howard at the Kitchen.

EARLY MUSICAL EXPERIENCES

Roads: Could you tell us about your early musical experiences and training?

Lewis: My father and mother liked music, though they weren't musicians and didn't know anybody who was. They suggested that I start to play something, and the trombone was my choice; I really don't know why. My father was trained as an electronics technician, so I used to get crazy lectures (with examples) about Ohm's law and Lionel Hampton. So you could say I'm my father's son.

I went to the Lab School in Chicago, a private school, so I had good teachers, mainly Dean Hey, a fine trombonist who was familiar with the contemporary, jazz, and classical literature; he also taught Ray Anderson. He and Frank Tirro pretty much ran the music department, and Ray and I were among only four trombonists, so we played in all the groups, the orchestra, the jazz band, the brass groups, and the concert band.

Mr. Hey started the jazz improvising classes about that time; he taught basic harmony and scales, and improvisation on chord progressions. He wanted the jazz band to have improvising soloists, and it was the school's best performing group. This was my first experience with music as a way for me to be creative. It was varied, but I never thought of being a musician or composer until I returned to Chicago after being away at Yale. I realized that there were all these people in the AACM [Association for the Advancement of Creative Musicians] around.

AACM WORK

Roads: Was the AACM led by Muhal Richard Abrams?

Lewis: Muhal is one of the founders, and certainly one of its strongest personalities, someone who can show you your interests before you know about them yourself. He took time to show me the scores of Stravinsky and the Second Viennese School. One of his big favorites is Morton Subotnick, who everyone got to hear if they went to his music classes. He is largely, and proudly, self-taught as a composer, and that philosophy of self-teaching has a large impact on the organization.

The AACM is a varied group of people. It includes gospel music, blues, funk, rock, classical music, jazz, electronic music, salsa, and contemporary music. I really should be putting all of these labels in quotation marks—they're sort of leftovers.

Roads: In what context did you play with AACM members as a musician?

Lewis: In all of those contexts, because we had gigs doing all of those things. Though composers seem to provide a focus to the AACM's activities, there are also sculptors, painters, dancers, programmers, and other visual, literary, and performing artists.

Roads: How long did you work with the AACM?

Lewis: It's an ongoing association. You don't really "work with" the AACM, you take part in its activities. The AACM sponsors concerts by its members and others, and you teach in its music school, which has a sort of "little red schoolhouse" approach—seven-year-olds going to school with seventy-year-olds. The more advanced students have to teach the beginners, so no one gets left behind.

In the school, it was always assumed that improvising was important, but they never taught it—maybe because they assumed you would learn that anyway if you were around them enough. You were always encouraged to become a composer. The members themselves would play your music, so I got quite a bit of practical information about instrumentation, orchestration, and the fundamentals of ear training, theory, harmony.

The AACM has always been primarily a composer's organization—like say, Composers' Forum in New York—except that it also addresses some of the specific problems that black composers and musicians have in pursuing their careers.

The AACM has always had two main aims, which its members feel are related: promoting new music activities and community involvement. The AACM's immediate community is the South Side of Chicago, where I'm from. There are parts of the South Side where people have money, but a lot of it is quite poor, particularly in cultural activities. So for people on the South Side, the AACM is important as a path to current musical thinking.

Roads: But you left Chicago eventually.

Lewis: Yes, about 1977. Quite a few of my musician friends had left for New York, even Muhal. I had started listening to people like Gordon Mumma and Robert Ashley, and in New York I was performing with Anthony Braxton, who introduced me to Richard Teitelbaum. Richard and Anthony put me on the path of actually being involved in new music, instead of just listening to it at home, which was mostly what I seemed to be doing in Chicago.

Richard in particular got me started in actually performing with electronic instruments, though I had to give up playing the synthesizer; maybe Charlie Parker had the same effect on saxophonists. Anthony and Richard are able to deal easily with integrating performers' ideas into a compositional framework. To accomplish this, you must have composer/performers, who understand about form and control as well as individual initiative. Richard's synthesizer playing is certainly a model for what I would like my interactive systems to sound like.

MICROCOMPUTER MUSIC

Roads: What happened after you arrived in New York?

Lewis: I met a lot of the people I'd been wanting to talk to. I was able to be involved in so many different kinds of musical activities. This is more possible in New York than in most places, certainly more than in Paris, where musicians seem more specialized. I was able to work with my old Yale classmate, Anthony Davis, whose work successfully integrates formal concerns with improvisatory freedom.

I had a fair amount of contact with both the "downtown" and "uptown" music scenes, and some contact with that branch of the avant-garde rockers who employed improvising in their work, such as Bill Laswell and Fred Frith. I also took part in the end of the "loft jazz" scene. The loft jazz places closed, but the "loft new music" places, such as the Kitchen and the Experimental Intermedia Foundation, continue. They have even expanded their programming to include music which they had perhaps previously thought of as being outside their realm. This idea of expanded programming has been quite a healthy development in the music scene in New York lately.

Roads: When did you begin to work with microcomputers?

Lewis: Jacques Bekaert, a composer and journalist, organized a concert with Rae Imamura, Douglas Ewart, and me at Mills College, around 1978. The scene at Mills seemed worlds away from the electronic music studios I had been exposed to. They still had the public access studio going at that time, and they let me try out the electronic equipment myself and showed me how things worked. Jacques introduced me to David Behrman, whose work is very important in the area of interactive performance with computers. David was rehearsing with Rich Gold, John Bischoff, and Jim Horton, who were using tiny computers called KIMs. They were not exactly my image of what computers were like—a board about the size of a sheet of paper with a tiny keypad and a few chips.

Roads: This group was the League of Automatic Music Composers?

Lewis: Yes. Each KIM was connected to a sound generation device, and all of the KIMs were interconnected. Musical data was sent between all the systems. Then, the four composers listened to the output of the machines. Occasionally somebody would halt his program to try a new value in memory or maybe jiggle a wire or something.

Roads: How did it sound to you?

Lewis: It sounded a lot like a band of improvising musicians. You could hear the communication between the machines as they would start, stop, and change musical direction. Each program had its own way of playing. I hadn't heard much computer music at the time, but every piece I had heard was either for tape or for tape and people, and of course none of them sounded anything like this. I felt like playing, too, to see whether I could understand what these machines were saying. I got a KIM as soon as I got back to New York and started trying to learn how to make assembly language programs, cheap digital-to-analog converters, and some other electronic doodads so that I could use the KIM with my synthesizer. But I wanted to play, too, so I had to find out something about getting my sound into the computer.

INTERACTIVE PROGRAMS

Roads: So from the beginning of your work with computers you've been involved with interactive programs—programs that interact with a performer.

Lewis: Yes, that's the only thing I've tried to do with a computer.

Roads: How are microcomputers suited to this task?

Lewis: Having your own machine means that you don't have to be tied to a large institution or have a lot of money. And as it turns out, the microcomputer people have explored some areas that are quite different from those studied at the large institutions. That was the interesting thing about David Behrman's programs. You could play beautiful melodies, and they would answer with something that was related to what you were doing. They were interactive. They didn't just respond to input with a predictable transformation. They were very simple, really, but extremely effective.

Portability is also a powerful consideration, particularly for the performing composer. Portability means that more people get to hear your music. Of course a tape is quite portable, but so far tape music hasn't figured as prominently in my musical thinking.

Roads: For someone working with real-time input, what special problems do microcomputers pose?

Lewis: Microcomputers, particularly the eight-bit machines, pose problems for a software composer/performer, because once you commit to writing programs as complex as mine are turning out to be, you have to fight for every microsecond if you want it to respond quickly enough to be interesting for the musicians who play with it. This means that you tend to look for the simplest solution that will yield musically interesting results. Every solution is evaluated in terms of memory overhead and computation time, as well as in terms of difficulty of coding, since I'm not a trained computer hacker. But apart from technical concerns, performance practice is my point of departure.

Roads: Could you elaborate?

Lewis: When I started thinking about computers, long before I could actually buy one, I was performing collective improvised music with an AACM group called Quadrisect, which included Douglas Ewart, James Johnson, and Mwata Bowden, all good woodwind players. We were attempting to make music by integrating scored material with improvisation.

You learn a great deal about human interaction in this way. You learn that everday interaction between people is largely improvisational on a moment-to-moment scale. When you take these everyday concepts into the musical realm, they become structural determinants for the music.

For example, in group improvisation, you always have teams, and these teams change throughout the performance. Erving Goffman points out that this happens in conversations among a group of people. For example, everyone might be trying to get one person to be quiet or to speak more, and there are all kinds of strategies for those behaviors. All these interactions can be translated into music. Linguistic interactions have been applied to music, but these conversational interactions have never been studied in a musical context.

Roads: So you're talking about a musical discourse theory. It could be modeled after the discourse theory of artificial intelligence (AI) research,

which studies the protocols needed to allow a machine to carry on an open-ended conversation.

Lewis: Yes. The study of improvisation in music should interest researchers working in AI, since many aspects of human information processing are at work on a moment-to-moment basis in an improvised music performance. So I thought it might be interesting to play music with a computer. Actually, I knew little about the problems involved when I started. I remember talking to Marvin Minsky and Maryanne Amacher once in Soho—and of course I had no idea who Marvin was at the time—and I said I was interested in buying a computer and building an interactive improvisation system with it. Well, you know Marvin; he never says a discouraging word. Everyone at the table seemed to think that it was a good idea that should be tried; we talked about it. It only dawned on me several years later what the whole discussion was really all about.

Anyway, when you look in the psychological or AI literature for information on improvisation, you don't find much. In the classical music literature, you find a fair amount of uninformed or culturally biased pronouncements, especially in the studies published in the 1950s and 60s, when Parker, Coleman, Taylor, and Coltrane were changing music just as Stockhausen, Cage, and Boulez were.

Books dealing with jazz improvisation are basically harmony manuals, though sometimes brilliant ones, like the work of George Russell, or sometimes they are collections of exercises dealing with chord progressions—what the old-timers called "riff books." The structure of the improviser's decision-making process has not been studied anywhere I've looked. Derek Bailey's book [*Musical Improvisation*, 1980, Prentice-Hall], though it doesn't discuss this topic either, stands out as the most interesting writing on improvisation that one can find at the moment.

IMPROVISATION TODAY

Roads: Since you have had extensive experience in improvised performance, how would you assess the place of improvisation in the contemporary music scene?

Lewis: I think it is a healthy situation overall. Many people are trying to learn to improvise, because it has traditionally been a wonderful way to learn about the possibilities of making music. There's a lot of interesting activity among improvisers—new methods of structuring, advanced ideas of how to integrate scores with improvisation, interesting new sounds, extended notions of what an "instrument" is, what a "virtuoso" is, what a performer's role is.

Today's performer/composers emerge from far more varied cultural and musical backgrounds, and they're talking to each other. The simple ideas that nonimprovising composers put forth on how to "use" improvisation as a technical facet of composition have long since been replaced by the notion of the costructuring that composers and improvisers can give to each other in a real collaboration.

At the same time, in that branch of contemporary music which considers itself to be an outgrowth of earlier European and Euro-American musics, the current fashion is anti-improvisation. This has been the case for over a century.

Roads: Why is that?

Lewis: Part of the problem contemporary composers have with improvisation has to do with the notion of a score as an artifact that, say, you leave to archaeologists as documentation of your work and of the cultural thrust of the period. Also, it's got something to do with the dislike some composers have of "losing control" of the composition, since it is true that when improvisation is part of the piece, you've just added extra composers to the mix. It is true that for certain kinds of structures, improvisation is not possible or desirable as a part of the piece. Of course, Bach, Beethoven, and many other past composers were quite good, even famous improvisers. However, the composer/performer started to lose ground in Western classical music after Beethoven's death, and in Richard Wagner, we had the prototype of the composer/conductor who does not possess concert-level instrumental technique. This leads us to the present-day situation, where relatively few composers have either interpretive or improvising skills, and today's absurd dichotomy between "composer" and "musician."

JAMMING AT IRCAM

Roads: You moved to Paris in 1982, and you have been doing some work at IRCAM. I understand that you recently had a computer music jam session with Salvatore Martirano and Donald Buchla. How did that come about?

Lewis: It was an impromptu session, with no audience except for David Wessel and a couple of other people. We were all working in Studio 5 on our own projects, and we just decided to see if we could hook up all the machines and play together. Studio 5 is central in IRCAM, so it became known as a place where strange things were happening—people drinking wine and jamming on computers. Sometimes David Wessel would come in and play drums.

I have always been interested in Sal Martirano's Sal-Mar Construction because it was the first machine I heard for automatic music that could change direction—rhythm, tempo, and spatial direction—in an amazing way. When

Sal plays it, or rather plays with it as I now understand, it sounds even better, but the machine itself is playing at a pretty high level. The Sal-Mar has a wonderful sense of pacing and phrase.

Sal has many interesting ideas about how to make musical patterns based on logic functions. What I like also is that he has a way of discussing his ideas that lets you know that the search for the right algorithm is made gradually by listening, by trial and error. You have to constantly listen to the phrasing. I used to listen to my program all day and half the night. I would turn it on for 10 minutes here, an hour there, and see what it would do. It's like listening to a musician practice, and every so often you say, "No, do it *this* way."

When I heard that Sal was going to be at IRCAM at the same time as me, I was anxious to see what would happen if we could get our systems to talk to one another. As it happened, we didn't have enough time to connect the output from my machine to his machine.

Roads: So the Sal-Mar Construction couldn't listen to your performance.

Lewis: Well, Sal started playing, and then Donald Buchla came in and started playing, as he had just brought in his new 400 series synthesizer. This really raised the ante, since Don can play quite ferociously at times. Both he and Sal could hear my machine, and I had inputs from both of them. So it was quite a session. I must say, my poor little system nearly got lost in the midst of all that intensity, but I learned more about how a band of interactive computer systems could sound.

PROGRAMS THAT LISTEN

Roads: One of the trademarks of your computer work is the creation of programs that model listening processes.

Lewis: Modeling a human listener is part of it. You see, I don't like playing solo trombone concerts. I would rather play with someone else. I wanted to make a program to see what sorts of things an improviser has to know in order to play successfully with another improviser. A key question has to be: How do improvisers structure their listening? You can't hear everything.

At any given point in the music, certain factors stand out. Understanding when and why those factors stand out is crucial to making a program that could hear a sound in the same way that an improvising musician might hear it.

Roads: But you begin with a head start in that you do not have to account for every detail of the music, but only the musically relevant cues. By contrast, in software systems for music transcription from sound to score, computer scientists are trying to write programs that hear virtually everything.

Lewis: When I first came to Paris, David Wessel got me together with Bernard Mont-Reynaud, who heads a research team at Stanford that is pursuing automatic music transcription work. His paper on the subject was quite interesting for me. Bernard is very wide-ranging in his musical interests, and can play, too. This is very important, because practicing performers bring a certain insight to this subject that a nonplayer might not have, like I was saying about Sal. It seems as though Bernard is trying to isolate some of the same musically interesting structures that I am, though I branch off by trying to make the program relate what it's hearing to an ongoing piece of music that it's also helping to create.

In performance, musical decision-making is much more immediate than it is in traditional composing. Many snap judgments are made. Some kind of context control is necessary, and I'm trying to help my machines understand musical context. Since good improvisers can't listen to everything, they have to keep track of the context in which they place the sounds they're making and hearing. You have to find the structure in what you've just played and heard or, if necessary, posit it or another structure as a point of departure.

Improvisers often work in terms of rather loosely defined "shapes," which can be defined in terms of characteristics such as volume direction, pitch direction, duration, rhythm regularity, pitch or duration transposition, time between major changes in output or input, pattern-finding, and frequency of silence. You don't need or want an exhaustive transcription, but instead a fast, general analysis of what's happening at any given moment and what's been happening. This requires massive, but musically important, data reductions.

With a small machine, the computation time required to do this means that you can't allocate most of the machine's computation time to the task of making a fancy sound. I'm not concerned about this, though, because wonderful FM [frequency-modulation] synthesizers with easy computer interfacing are available. So instead of worrying about the sound, I think about what the machine is going to play and how it's going to hear. For me, live electronic performance has less to do with so-called "interesting" timbres than with the directed quality that you find when a human is playing.

CHAMBER MUSIC FOR HUMANS AND NONHUMANS

Roads: I understand you have been working on a program called Chamber Music for Humans and Nonhumans. Could you describe it?

Lewis: This is really the only musical program I've ever worked on, though I've used parts of it to realize simpler pieces, such as my new *Songs*, which are interactive performances of some of my miscellaneous melodies, along with some transcriptions of themes by John Coltrane for computer/

human performance. Over the years the old program has improved and reached a point where it plays fairly well. People don't mind performing with it.

Roads: When did you start to invite other improvisers to interact with your program?

Lewis: Right from the beginning. It's an important part of my work. I'm interested in their comments on how it plays. My colleague Douglas Ewart has played with all the systems, and he'll be part of the concert I'm going to do at IRCAM in 1984, along with Steve Lacy, Joelle Leandre, and Derek Bailey. Earl Howard has played alto saxophone with it, among other performers.

In the program I attempt to distill the knowledge I've gained from being involved in improvising. I can hear the effect of this distillation in the program's output, as can any listener. The next problem for me is how to make the machine remember and reuse the things it finds in its little world.

Within the program there are certain things that could be thought of as performance techniques. Because of this, I'm starting to think that the computer could play a regular composition along with human musicians. Of course it would be a composition which involves improvising, though not as a "technique" of composition. Treating composition or improvisation as a technique cheapens its possibilities.

Roads: What is your definition of a technique?

Lewis: To me, the central activity of music-making is context control; some people call this "form." Techniques are like bodies of knowledge (and I include motor skills in this) that can be useful for manipulating contexts, or for making sounds that are useful in a given context, or for realizing a certain set of sounds. To control the way in which a sound relates to other sounds, we need music-makers. Thus improvising and composing are not things you use, but creative activities, by definition controllers of context.

Roads: How do composed and improvised elements exist together in the same piece of music without interfering with one another?

Lewis: This is to me one of the most exciting problems. What is clear to me is that you have to plan a piece wherein the sounds that people play don't matter, or where they aren't always the first thing conceived. That reverses the traditional emphasis on pitch and timbre and brings out rhythm, pacing, and interaction strategies. It also makes necessary the idea that everyone performing the piece has increased responsibility for the piece's formal success.

Including a computer program as a costructurer of the music is a problem for some musicians. They feel they should somehow be controlling the machine, rather than working with it to make music. This attitude would obviously be unacceptable in an improvisation with a person. You have to listen to get anything out of an interaction.

You basically need a new kind of musician to do this kind of piece. Musicians have to know something about composing and improvising. Composer/performers are better at interpreting the pieces I make because they understand the interactive nature of composing and improvising—that everyone is a part of the same process of making and finding order in music.

MULTIPLE MICROCOMPUTERS

Roads: What is the basis of your current work at IRCAM?

Lewis: I'd like to make the system more responsive, have it look at more kinds of events, and place those events in more varied temporal contexts. I want it to remember more of what it has heard and look at events on a larger time frame, so that it can recognize sections within a piece and detect when a "start window" for a new section has occurred. I would also like it to be more independent, so that it is aware of when someone is trying to oppose it or harmonize with its playing. This means it must have some idea of when it's relating directly to its input and when it is going off on its own. I'm close to achieving most of this now. Later, there'll be a score that both the computers and the musicians can play together, passing cues back and forth directly through the music. I don't want the musicians looking at a video screen.

I've always worked with small computers, and even though I don't have to at IRCAM, I'm still using Apples—several of them. I think that enough of them together can do the job, and, as a practical matter, I can perform the work wherever such equipment is readily available. There's almost always a computer store in town, so it's often possible to arrange for a number of Apples to be brought to the performance space.

The present system is programmed in FORTH, an interactive language that is well suited to machines with limited memories. Perhaps in a later version I can take advantage of the newer personal computers and LISP-like languages.

In any case, the system I am building now includes three independent melodic players that share a common "ear."

Roads: So will the computers be listening in three different ways or generating sound in three different ways?

Lewis: Each player runs essentially the same program and listens to the same ear, so it is each players's experience in playing and hearing that makes it come into its own. Each of these players gains character by playing exclusively with a particular improviser for an extended period.

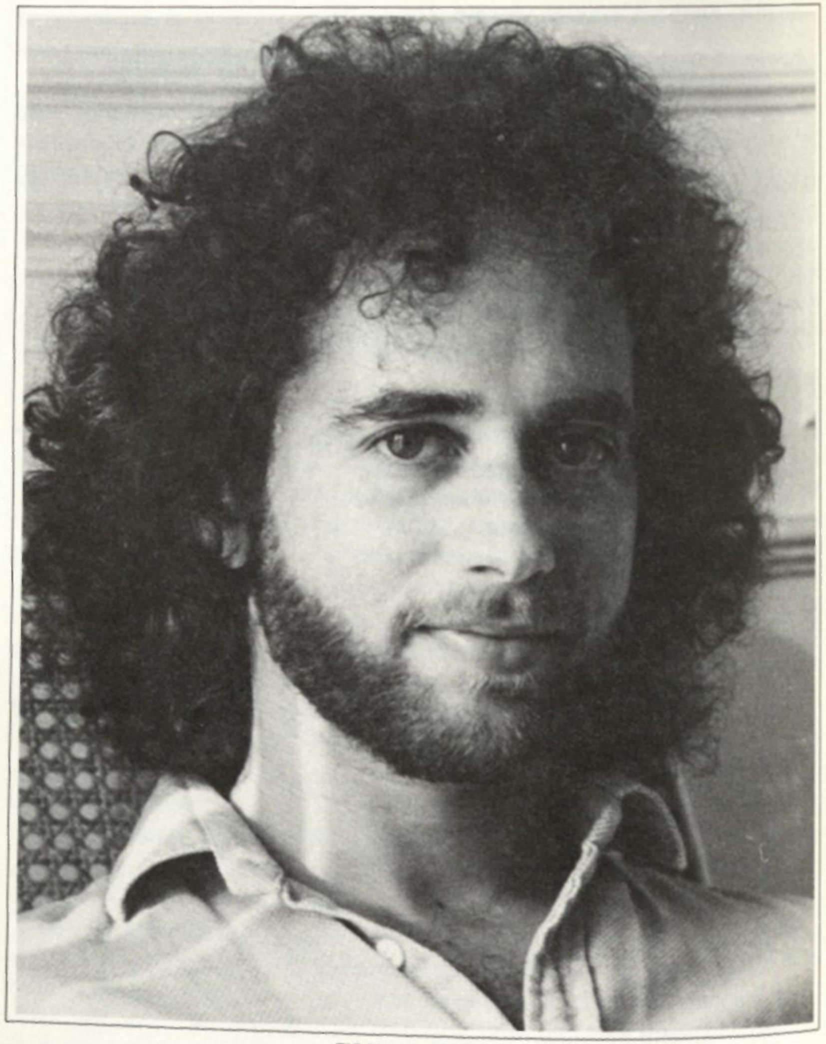
Roads: So is it a learning machine?

Lewis: I wouldn't want to say "learning;" it's more like adapting.

A PARTIAL DISCOGRAPHY OF GEORGE LEWIS

As composer:
Chicago Slow Dance: Lovely Music 1011
Douglas Ewart/George Lewis: Black Saint 0026
From Saxophone with Trombone, with Evan Parker: Incus
Homage to Charles Parker: Black Saint 0029
Yankees, with Derek Bailey and John Zorn: OAO 5006

As performer:
Laurie Anderson, *Big Science*: Warner 3674
Anthony Braxton, *The Montreux/Berlin Concerts*: Arista
Anthony Davis, *Hemispheres*: Gramavision 8303
Jill Kroesen, *I'm Not Afraid of Girls*: Antarctica 6201
Steve Lacy, *Prospectus*: Hat Hut 2001
David Murray Octet, *Home*: Black Saint BSR 0055
Material, *Memory Serves*: Celluloid 60042
Neil B. Rolnick, *Solos*: 1750 Arch Street
Mathias Ruegg, *From No Art to Mo-(z)-Art*: Moers Music
John Zorn, *Archery*: Parachute 17/18



TOD MACHOVER

6

THOUGHTS ON COMPUTER MUSIC COMPOSITION

Tod Machover

Music has changed fundamentally in this century. In spite of current trends toward nostalgia in contemporary music circles, musical expression does not turn back. Although composers are not in a position to be historians or too conscious determiners of their own fate, each individual must grapple with certain basic issues: Does music continue to play a major role in society? Is there a musical language that will clarify underlying similarities between seemingly diverse materials? What forms and structures convey most effectively the complex relationships that exist among these new materials? One could go on and on.

Work with computers exaggerates these problems and forces the composer to find at least provisional solutions to them while simultaneously conducting creative work. This is due to three major factors. First, the computer offers powerful possibilities for constructing a new sound world (far exceeding those offered by traditional instruments or analog electronic means) and for controlling with the greatest care and precision the minutiae, the atomic structure, of sounds themselves. Second, the computer suggests new ways to think about musical structure because of the unprecedented facility for unifying macro- and microlevels of a composition. The machine gives the composer the capability of applying analytical and theoretical concepts expressed as compositional algorithms or programs, prompted by the necessity of organizing the new sound world that has become available. Third, by establishing an interaction between the composer and technology, the computer stimulates thought about the compositional process itself and suggests a new relationship between creator and material with the computer functioning as a more or less active intermediary.

The computer stimulates the imagination and provokes thought about major questions; it is not in itself a store of answers. It imposes no aesthetic or theoretical constraints and fewer and fewer technical ones. It exists as a thinking tool that forces composers to look at it and to look even harder back at themselves to invent limits and rules and to imaginatively create music with them.

All this is no small task. More than ever before, the composer is asked to play many roles simultaneously: researcher, instrument builder, performer, theoretician, as well as creator. It is my strong belief that involvement in these tasks can shed light on compositional problems and can constantly stimulate the

imagination. The composer can never forget, however, that the most important process is always one of intuition and judgment (often based on "insufficient evidence"). No matter how extensively the composer engages in rigorous research, confidence should never be lost in the power of simple musical thinking.

To convey my thoughts about basic questions of musical language and structure and to show the influence of the computer on this development, I will briefly discuss three compositions of mine in this article: *Déplacements* (1979), *Light* (1979), and *Soft Morning, City!* (1980). Each piece uses some form of computer-generated tape in combination with live instruments. A variety of different techniques (technological and compositional) are represented. All these techniques are concerned with the relationship between instruments and electronics, and they involve the juxtaposition of many contrasting elements. Each treats this in a different way. In the music, formal principles hinge on the balance between these complex paradoxes and revelations of unifying principles that connect them.

DÉPLACEMENTS

Déplacements emphasizes the differences between a live instrumentalist, in this case a guitarist, and an artificial world existing on two-track tape. The electronic score, based on analyses of the guitar spectrum, hardly resembles the guitar part in timbre, articulation, or structure. This electronic part establishes a set of rules that produce results that seem static on the surface but quick-changing and intricate on closer listening. In fact, the conceptual model for this computer score was provided by the sounds that one hears in the inner ear, always present but made up of infinitely differentiated, scarcely perceptible details. In this work, the idea was to juxtapose the calm, introverted, unyielding tape with the solo guitar part, which becomes more passionate the more it tries to calm itself by matching the tape.

Both parts are based on the same collection of structural pitches. However, the tape uses them as a series of objects to be looked at with a structural "microscope," to be entered into and explored, whereas in the guitar part these pitches provide the basis for long-range harmonic and melodic relationships.

This electronic part was created with Giuseppe diGiugno's 4C machine (Moorer et al. 1979), an extremely powerful digital synthesizer constructed at IRCAM (and part of a series of which the 4A is the predecessor and the 4X is the final prototype). The electronic part of *Déplacements* was the first tape to be produced on the 4C. The approach to this machine was to design instruments, write scores (either traditional note lists or small compositional algorithms) and reserve control over certain parameters for real time (either in rehearsal or recording stages). This relationship between predetermined and real-time control varied depending on the instrument and, in certain cases, the section of the composition.

Some examples of the processes used will now be described. The durations, amplitude envelopes, and speaker placements for a complex random-noise generator were specified in a score. The bandwidth and center frequency of a band-pass filter were controlled in real time with potentiometers. A double frequency modulation (FM) instrument also chose duration and center frequency from a score and was used to generate slowly varying complex timbres. The index level of this FM instrument was controlled by pots, although these were multiplied against complex functions programmed in the score. In addition, a series of *c*:*m* (carrier-to-modulator) ratios were stored in memory, and these were chosen in real time by buttons, thus allowing real-time control over both the amount of modulation and the spectral content. Another, simpler FM instrument had its *c*:*m* ratio, center frequency, speaker movement, and other characteristics preprogrammed.

These passages were generated by an algorithm. The overall form was determined in the program, and intermediate notes were chosen by a constraint system; real-time control allowed for the selection of various available processes. These compositional algorithms, although suitable for the restricted purposes of *Déplacements*, are very simple. Much work needs to be done on LISP-like editing languages to turn these types of procedures into powerful compositional tools.

These elements, along with many others, were broken down so that each could be regulated and "performed" carefully. Many such voices were superimposed using a 16-track tape recorder. As mentioned earlier, combining the different layers creates transformations and variants—mostly compressions or elongations in frequency and time—of analyzed guitar spectra. The combination of so many complex timbres, each performed with its own repertoire of controlled musical parameters, gives a result that would be possible only with a computer—and, perhaps, only with one that generates sound in real time.

This is not the place for an exhaustive discussion of real-time composition. However, I would like at least to mention my belief that the method of combining the power, speed, and precision of computer calculation with direct, intuitive interaction on the part of a musician seems indispensable to all future development in computer music. One can replace overcomplicated and abstract instrument designs with simpler ones that allow for direct control of musical results by live input devices. We are at the beginning of real-time work, and enormous progress needs to be made in designing control instruments that can carefully monitor subtle hand and body movements of a performer and translate them into musical gestures. We need sophisticated graphic aids (including notation systems that represent new thinking about structures and sound materials), and ever-more-powerful and elegant programming aids that allow the composer to efficiently control and organize musical ideas. And we need to create standardized, concertworthy performance machines that are easily transportable. A machine such as diGiugno's 4X, currently in operation at IRCAM, has already convinced me of the enormous potential that lies waiting to be exploited by imaginative composers (Asta et al. 1980).

LIGHT

The principle of juxtaposing different musical layers is extended in *Light*. Three main forces are contrasted: a group of 14 live instrumentalists and two four-track computer-generated tapes (Tape I and Tape II). The two sets of four speakers are widely separated in the performance space, and the instrumental group is divided into four subgroups that are placed in the four corners of the hall.

Corresponding to these groups, four related but quite different subsets of the musical language of the work are established, one fundamentally linear and melodic, one harmonic, a third rhythmic, and a fourth timbral. In addition, two types of electronic material develop, both extensions of the instrumental world. One uses synthetic models of each of the instrumental groups and goes beyond the limits of human playing in many ways. This tape is closely synchronized with the instrumental part and is divided into short segments that are seldom longer than 20 sec. It is controlled by a performer who starts and stops a tape recorder with ensemble-playing accuracy. The second tape moves further in the direction of the electronic sound world found in *Déplacements*. There is a constant shift between dense static textures and bursts of flickering detail. This tape is rarely coordinated with the rhythmic movement of the instruments and runs almost continuously throughout the work.

These two electronic sound worlds called for many different composing techniques, notational systems, and computer resources. Tape I was realized using the 4C machine, which is ideally suited for careful and subtle tuning of many parameters in real time, but has limitations (only 64 oscillators, 64 multipliers, 32 envelopes, etc.) that make it difficult to produce additive synthesis. This is better achieved with the 4A machine, which, with its bank of 256 real-time oscillators, was the perfect tool for producing Tape II.

It is not possible to discuss here all the synthesis techniques used in *Light* or to indicate how all the materials are organized. However, two short examples might help to indicate some of my thinking in this work and to show why the computer plays such an integral part in it.

Tape I

For Tape I, I began by constructing a set of synthetic instruments that related in timbre and articulation to the live instrumental groups. Each used a different synthesis technique:

- A family of brass instruments using nonlinear synthesis
- String instruments with complex-carrier FM
- A clarinet/oboe with additive synthesis
- A piano with wavetable synthesis

My interest in creating these synthetic materials was not to mimic or equal live instruments (such an effort being problematic technologically and of doubtful interest musically). Rather, I sought resources that could be used as starting points for far-reaching transformations but would have strong common bonds with real instruments. The computer, by providing access to the fundamental building blocks of separately controllable harmonics, gives the composer an unprecedented power to control timbre as an indispensable compositional parameter.

Each of the four computer instruments was given a characteristic library of transformations, and a fifth instrument was created to allow for interpolations between the four timbral states and for a wide range of complex nonharmonic spectra as well. All these features were used to emphasize structural aspects of the music and to create juxtapositions with and elaborations of the live instruments.

For instance, at one point in *Light*, a synthetic piano and a live piano are put in opposition to each other. As just mentioned, the synthetic piano uses a repertoire of wavetables representing an averaged power spectrum of piano notes analyzed in different registers of the instrument's range. The passage begins with a normal spectral distribution but gradually substitutes wavetables of low piano notes for notes in the middle and upper registers, producing spectral contortions. As the synthetic piano ascends in frequency, the timbre is transformed more and more, but "correct" spectra are reserved to highlight a major melodic motif that recurs at this moment and to emphasize the harmonic material that becomes the basis for the next major section of the piece. This material, serving as an important structural transition, is picked up by the live piano, which reenters at this point, submerged in a timbrally deteriorating shadow of itself (fig. 6.1).

Tape II

The second tape for *Light* made no attempt to imitate live instruments; rather, it set about trying to shape a more idiomatic electronic environment. All the work

Fig. 6.1. Excerpt from *Light* (1979) by Tod Machover. (Used by permission of G. Ricordi Publishers, Paris.)

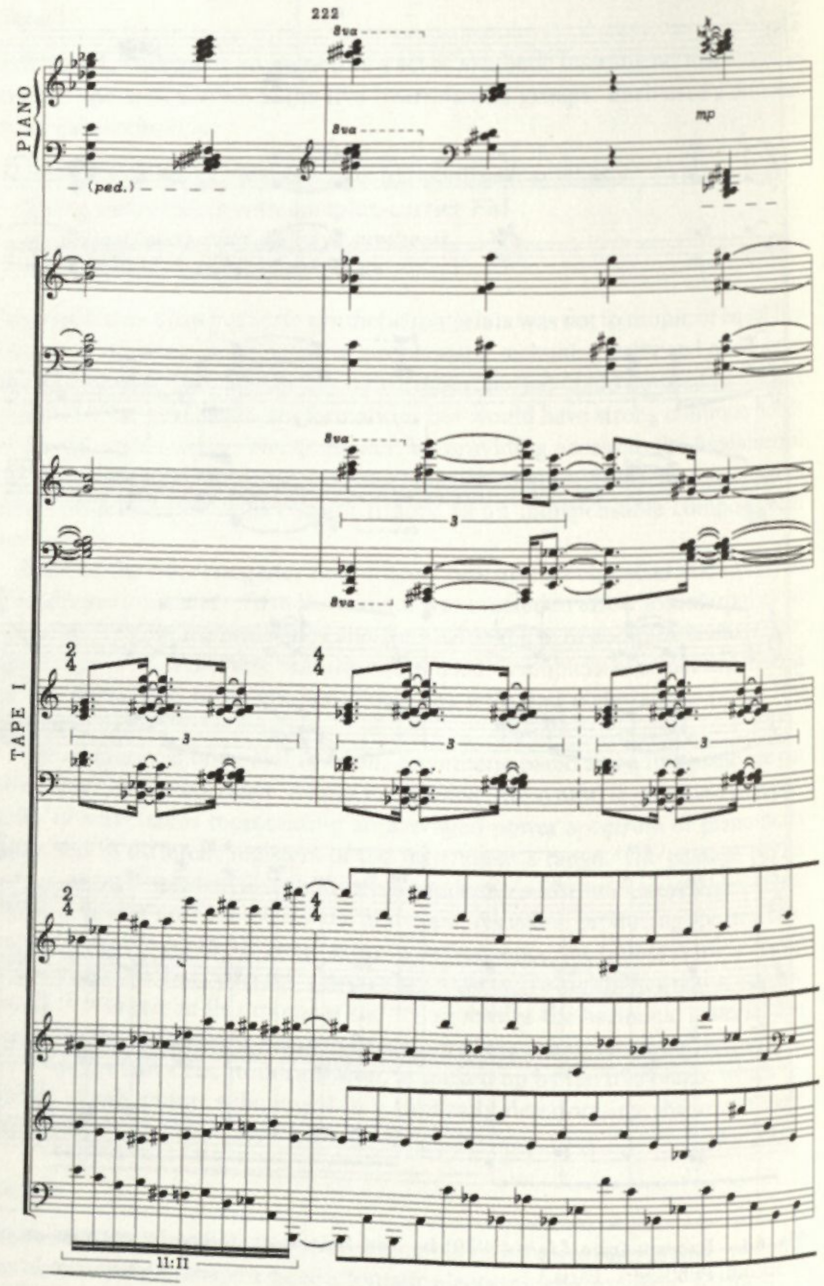


Fig. 6.1. *Continued*

Fig. 6.1. *Continued*

Fig. 6.1. *Continued*

for this tape was done on the 4A. Individual parameters were not controlled in real time, but many decisions about processes and structures were. It seems to me that the biggest challenge open to composers working in the electronic medium is to structure and limit the vast ocean of sound resources that do *not* resemble those of traditional instruments. This development will take years of hard work, and the computer will be an essential tool.

For *Light*, a system of related spectra were created that included a method of moving from one spectrum to another using common partials as pivots. Around these guideposts a set of variables were defined and controlled by a program: the frequency of each of the 256 oscillators, their rate of change, waveform (only one 16 Kword wavetable at a time with the 4A), and phase. In addition, certain more general variables such as density, sharpness of attack, random deviation, glissandi, chorus effects, and speaker placement were devised as weighting features to be included in the real-time programs controlled from a computer terminal.

A continuum was created that could transform any given sound through a variety of states from clear pitch to noise. For instance, an event could begin with all 256 oscillators grouped closely around a certain frequency, say 440 Hz, with a slight random deviation to establish an effect of a chorus of voices. A procedure generated in real time could gradually assign oscillators to each partial of the harmonic series based on a fundamental frequency of 440 Hz, and each of these partials could in turn be treated as the center frequency of a new harmonic complex.

Because of this spectral weighting, certain frequency regions soon become saturated and pass quickly from a predominant sensation of pitch to that of a time-varying harmonic spectrum, a complex chord, a cluster, and, if the saturation is great enough, to noise of a certain color. These changes were controlled on a local level to generate specific events and gestures, but they also determined the longer-range movement of the musical material. A 20-minute development of increasingly dense frequency, speed, attack envelope, harmonic motion, and homophonic synchrony was created. This process ends with the transformation of what at the beginning of the composition were vaporous, delicate noises into thunderous, climactic crashes, as if tiny particles had formed into matter in front of one's eyes (ears!).

The system of organizing complex spectra and of creating a pitch-noise continuum was largely intuitive in *Light*, and it would be difficult to demonstrate clearly the theoretical base on which these decisions were made. Much more work on the part of composers, theoreticians, psychoacousticians, and others in the computer music field needs to be done before a coherent set of rules is developed to define these phenomena.

SOFT MORNING, CITY!

Soft Morning, City! provides an interesting contrast with the two works discussed so far. Although many concepts and materials are similar in all three pieces, this last work attempts an integration between the tape and two live performers (soprano and double bass) and strives to achieve a kind of chamber music relationship.

The work of James Joyce is a strong interest of mine. It had long been my wish to set to music the final monologue from *Finnegans Wake*. This passage is a supreme example of the unity of verbal meaning with the rich polyphony of musical sound, weaving many layers (reflections about the past, philosophical asides, sounds themselves, and infinitely more) into a unified fabric. I imagined the soprano Anna Livia Plurabelle as the center of the composition. All material stems from the vocal part, and the double bass and computer form a sort of musical aura around it. Digital electronics made it possible to emphasize certain words, create superpositions, disembody the soprano voice—in short, to create a large extension of the soprano presence. Thoughts originate with Anna Livia; the computer becomes the resonance of these thoughts in her own mind or in the river Liffey to which she speaks and sings.

There are so many techniques for creating and transforming vocal sounds used in *Soft Morning, City!* that I can only touch on them here; it bears repeating that few of these techniques would have been realizable without the use of a computer. I found it necessary to combine the resources of the two digital synthesizers, the 4A and the 4C, with those of the Digital Equipment Corporation PDP-10 computer system at IRCAM. Although using these machines together in one composition was not very convenient at that time, their complementary dissimilarities made the combination a very rich one. Techniques were used to highlight the meanings of words and to create a sense of continuity and layers of musical meaning, based on these words but independent of them. Four types of transformation are most prominent:

- The treatment of entire words, mostly prerecorded from a live soprano and transformed by the computer
- The use of microediting to give vowel or consonant sounds a strong presence
- Transitions from vocal sounds to those of the double bass or to nonharmonic spectra, as well as cross-synthesis with the bass
- The complex combination of many techniques, either on the same initial word or sound or on contrasting but related groups of material

The words that are transformed electronically are often those that Joyce himself reiterates, such as "time, peace, remember, soft, again, come," and so on.

In general, each word is treated with a characteristic transformation that identifies it when it occurs in dense contexts and when it returns in the composition.

A typical example is the phrase "Time after time . . .," which is stated first by the soprano and then on the tape (the soprano moving on to other text while the phrase is electronically elaborated). The word "time" was recorded from a soprano and digitized. It was then run through a series of linear prediction programs implemented by James A. Moorer (1978), which analyzed the formant structure of the word and created a set of time-varying filters to represent it. These filters were then used as a model for resynthesis. The articulation and phrasing of the original word were left intact, but the pitch, timbre, and timing were all changed. The timbre became a purely electronic sound, with the initial "t" of the word greatly emphasized. This new sound was produced 12 times, each at a different pitch level. These in turn overlapped to produce the 12-note chord that forms the harmonic basis for that particular section of the work. Simultaneously the words "time after time" are superimposed, producing an initial "time after . . ." followed by a very dense mixture of the consonants "t" and "m" and the entire phrase emerging at the end. This resynthesized singing and the word jumble are added together. The entire fragment, lasting 10 sec, conveys the desired sensation of a musically intensified concept—time repeating endlessly (see fig. 6.2).

A similar procedure is performed on the word "come" in the same section of the piece. Here the timbre used for resynthesis is very complex, starting with a simple spectrum but evolving through five attacks to a sort of string instrument timbre. The final note on G is distinctly a string sound, still articulating the word "come"; the word grows out of the voice to play a quintuplet at the end of the measure and then transforms a third time to form the basis of the nonharmonic spectrum that characterizes the next event.

Sonic organization is of primary importance to Joyce in *Finnegans Wake*—the meaning is conveyed by the rise and fall of consonants, vowels, phrases, sentences, and whole passages as much as by the words themselves. Joyce has taken the trouble to create families of sounds that are traceable to specific core words, such as the river "Liffey" (Lst. . . lft. . . liv. . . leafy. . ., etc.) and "soft" (sft. . . sim. . . som. . . sm. . ., etc.).

I developed this tendency further and used sonic patterns to emphasize long-range musical movement. Sounds were created either by treating recorded examples or by creating synthetic ones. I used the computer's ability to edit sounds at the most minute level—down to the millisecond. For instance, 25 msec were taken from the beginning of the word "soft" and multiplied by a smoothing function to eliminate clicks caused by discontinuities in the waveform. The short "s" sound was then mixed with itself in a sort of endless loop to give a rising and falling cadence of "s" sounds; it was analyzed and

Fig. 6.2. Excerpt from *Soft Morning, City!* (1980) by Tod Machover, showing variations on the word "time." (Used by permission of G. Ricordi Publishers, Paris.)

Fig. 6.2. *Continued*

resynthesized to create great elongations in time. Then it was augmented in intensity using reinforced spectral components, and these various results produced a library of sounds based on the letter "s."

These procedures were performed on approximately 25 different sound patterns from the text. Computer mixing in combination with abstract electronic reinforcement allowed transitions to be made from one sound to another. Vowels provided even more possibilities than consonants, because considerable research has been done in creating synthetic vowel sounds.

In addition to techniques similar to those just described, programs developed by John Chowning at IRCAM in 1979 were used as models of singing voices (Chowning 1980). These programs allowed for transitions between different vowel sounds and for various other timbral shifts. They were combined with little slivers of edited vowels to produce results that clearly resembled the voice but were deprived of all emotional or expressive content, conveying an angelic, unearthly feeling in contrast to the live voice.

Great care was taken to use the computer as a mediator between the soprano and the double bass. At certain times, the natural opposition of the two was exaggerated by reinforcing the low end of the bass spectrum while creating a sort of inharmonic "glow" on the upper end of the voice spectrum. On the other hand, the difference between these two instruments was minimized at certain points in the work. One method used was to superimpose a bass-type sound (either synthesized or recorded) with an enriched version of the same spectrum created by additive synthesis. By gradually interpolating this synthetic spectrum to resemble vocal formants, a transition could be made to a real vocal sound.

The cross-synthesis technique was used extensively, although usually in relatively nonstandard ways. This technique, which cannot be discussed in detail here, involves the analysis of a recorded vocal sample, usually of speech, and the substitution of a nonvocal source to excite (glottislike) the filters set up by this analysis. This often creates a sort of "singing instrument," an effect that does not interest me. Rather, I saw the technique as a powerful tool for creating transitions between different timbral areas, for organizing a continuum of intelligibility of text, and for establishing a nonidentifiable hybrid that has feet in the worlds of both voice and bass but is quite different from either.

A major compositional transition uses this method, about one-quarter of the way through the piece. The opening section, where an intelligible voice grows slowly out of dense, noiselike structures, centers on a delicate tracing of word imagery. The double bass acts as a melodic and harmonic shadow of material stated by the soprano. The following bridge introduces a second major section where this bass-soprano relationship becomes more explicit.

The words "Soft morning . . ." were recorded by the soprano and edited into many smaller subunits ("s . . . sof . . . oft . . . m . . . mor," etc.), each of which was run through the analysis part of the cross-synthesis programs. Material corresponding in length to the desired end product was recorded from the double bass—a complex passage of modulated noise from the instrument's upper register. This bass recording was then treated with a variety of processes: filtering, superposition, transposition, spectral enhancement, and complex reverberation. The result, although resembling the double bass in articulation and phrasing, had a much richer and more varied spectrum. This material was divided into smaller groups of varying lengths that were used as excitation sources for the vocal filters that had been stretched and contracted in time. These fragments were then mixed to form a very complex result; one barely perceives the words but somehow understands them and strongly identifies the various consonant and vowel sounds that are used throughout the work. At the end of this 20-sec passage a similar cross-synthesis of the word "yet" appears; the "t," greatly exaggerated, and this marriage between bass and voice (used here for the first time in the piece) form the basis for the following section.

Figure 6.3, the opening of a very dense climactic section in the middle of the piece, demonstrates how these various effects were combined in *Soft Morning, City!* In Joyce's text, complex narrative is simultaneously commented on, moral and philosophical principles are offered, and Joyce's "structural sounds" are reiterated. Although there is an enormous amount of text, I felt that compositionally none of it could be left out, a familiar historical problem similar to that which composers face, for instance when setting the Credo of the Mass (note Beethoven's solution in the *Missa Solemnis*). I solved the problem by regarding the text as true polyphony that offers simultaneous views and feelings of the same phenomena. The computer processes made possible the superposition of many layers without loss of clarity or definition.

Stave 1 of the computer part of figure 6.3 represents music derived from recorded sung material, which is amplified and extended using synthetic voices and microediting techniques. The second stave represents word transformations based on recorded spoken text processed by editing and spectral reinforcement; it establishes a wide continuum extending from exaggerated intelligibility to dark murmers hardly identifiable as human voices. The third stave contains double bass-related material, including filtering and ring modulation of the recorded bass and cross-synthesis with the voice, the latter used to convey the bulk of the long text. The fourth stave is the product of synthetic material created on the 4C machine that centers around a complex instrument transformed from a double bass sound to one that is densely inharmonic. (Six other lines, mostly of inharmonic reinforcement and enhancement, were

Fig. 6.3. The opening of a dense climactic section in the middle of *Soft Morning, City!* (1980) by Tod Machover. (Used by permission of G. Ricordi Publishers, Paris.)

Fig. 6.3. *Continued*

Fig. 6.3. *Continued*

created on the 4A machine and the PDP-10 computer and are not represented in the score.) Simultaneously, the live soprano establishes a continuity, although juxtaposed with a certain linear abruptness, linking all the modes of expression previously used in the composition. The soprano highlights the major points of meaning that are being commented on by the computer part. The double bass serves the same function for prominent harmonic and melodic material.

The whole section strives to achieve what is for me a major formal and expressive goal: the gradual saturation of the listener by the presentation of a very large amount of carefully related but independent details to the point where a change or leap in the perception process takes place; elements that are seemingly strongly opposed become fused into a higher level of meaning, and this new unified complex becomes the basis for further development, juxtaposition, and synthesis. This principle, plus those previously discussed (highlighting musical layers according to function and the establishment of large-scale structural evolution of sonic units and transformation continua), is fundamental to the unified construction of *Soft Morning, City!*

CONCLUSION

The topics presented here have arisen out of compositional needs in my own work. They are held together by a developmental logic that is as much a product of intuition as of preordered searching. Two overriding principles recur most often and are integrally related to computer possibilities. One such principle is the use of the computer to create two completely different worlds, at polar opposites from each other, that serve as extensions (infinite extensions perhaps) of traditional concepts of instrumental music—that is, the creation of a totally abstract electronic world on one hand and on the other a sort of "superorchestra" of synthetic instruments that resemble traditional ones but far surpass them in various ways. The second principle is the exploration of the pitch-timbre-noise triangle and the fact that, in a fundamental sense, the three are simply special cases of a single phenomenon, controllable by the same principles and with relative ease (in programming, if not conceptually) on the computer. Both areas demand extensive development in the future, especially in their compositional and theoretical aspects, and await the establishment of organizational and structural rules of a greater generality than those developed so far. The use of the computer to produce complex compositional procedures and algorithms is certain to be an integral part of this work.

Real-time synthesis has been a constant interest of mine, and the works described here have been linked strongly to the development of IRCAM's digital synthesizers. The combination of such a machine as diGiugno's 4X with powerful real-time graphic tools, a composition editor/processor such as a modified LISP machine, a family of performable input instruments, along with a smaller, concert-size version of the same system, is a dream that is ambitious, very tempting, but not unrealizable in the coming decade. In the not-too-distant future one can imagine machines of similar power being connected to home computers to provide a much-needed communication of musical ideas and participation in musical culture among large numbers of people.

Perhaps it should be stressed once more, however, that no techniques, either of sound transformation or of structural organization, are of any interest whatsoever if they do not stem from deeply felt and highly logical musical thoughts. This process is of course symbiotic: musical visions shape the techniques and materials that are developed with computers while, at the same time, the musical imagination is stimulated by familiarity with possibilities that already exist. Neither process is better than the other and, at least from my experience, both are integral to effective computer music work, especially until that time when a more universal language of description and analysis is developed. The composer has to guard against being seduced by the infinite resources that

become available with computers and must accept the reality that it is in the processes of limiting and organizing that valid music is generated.

We are at the frontiers of a new era in music, one that will offer fresh and exciting answers to fundamental questions of language and organization, propose a clearer role for music in the social structure and a surer form of communication to a public, and integrate and express that which is most important about life as it is now and how it should or could be in future generations. We are only at the beginning of this road, so it is above all a time for composers to maintain a certain humility at the immensity of the task to be done. We must realize the need for common effort but simultaneously keep the courage and vision necessary to trust our own judgments, thoughts, feelings, and, above all, *ears*, and do what real composers have always done: stop talking and write music!

ACKNOWLEDGMENTS

I wish to express my extreme gratitude to Giuseppe diGiugno and Jean Kott for their patient collaboration and aid with the compositions discussed in this article, to Pierre Boulez for offering me the opportunity to work at IRCAM, and to all those on the staff there, without whose support none of this work would have been possible.

REFERENCES

Asta, V., A. Chaveau, G. diGiugno, and J. Kott. 1980. The real-time digital synthesis system 4X. *Automazione e Strumentazione* 28(2): 119–33.

Chowning, J. 1980. The synthesis of the singing voice. In J. Sundberg and E. Jansson, eds. *Sound generation in winds, strings, and computers*, 4-14. Stockholm: Royal Swedish Academy of Music.

Moorer, J. A. 1978. The use of linear prediction of speech in computer music applications. *Journal of the Audio Engineering Society* 27(3): 134–40.

Moorer, J. A., A. Chaveau, C. Abbott, P. Eastty, and J. Lawson. 1979. The 4C machine. *Computer Music Journal* 3(3): 16-24. Reprinted in Curtis Roads and John Strawn, eds. 1985. *Foundations of computer music*. Cambridge: MIT Press.

JEAN-CLAUDE RISSET

7

DIGITAL TECHNIQUES AND SOUND STRUCTURE IN MUSIC

Jean-Claude Risset

"I dream of instruments obedient to my thought and which with their contribution of a whole new world of unsuspected sounds, will lend themselves to the exigencies of my inner rhythm."

Edgard Varèse (from 391, no. 5, June 1917; translated from the French by Louise Varèse)

For a long time, Western music has evolved to include more and more complex relationships between musical elements. This evolution stretches from the birth of polyphony to contemporary serialism and stochastic composition. One may notice, however, that the increase in complexity of the relationships between units has occurred at the expense of the complexity of the units themselves. Polyphony demanded that individual lines give up much of the melodic microvariations that are outstanding in the music of some Eastern civilizations. The equal-tempered scale opened rich possibilities for traveling among tonalities—at the price of a compromise in tuning accuracy. Complex contemporary methods for composition still use mostly traditional instruments played in equal temperament.

The computer seems the ideal tool to amplify this relational complexity one step further, thereby enhancing the Western trend toward formal, hierarchical, combinatorial processing. Yet for some composers, myself included, the computer is more appealing as a vehicle for exploring another avenue, namely the elaboration of sound structure in music (Erickson 1975).

Until Debussy, sound structure was rarely in the foreground of Western music. Varèse shifted the emphasis from the organization of musical superstructures to the organization of the sound itself; the compositions of Cage and Ligeti often focus on the elaboration of the sound. New digital synthesis and processing of sound can offer sound material of unprecedented ductility, bringing us closer to a capability that Berio and others describe as: not only composing with sounds, but composing the sounds themselves. Digital processes make possible such a refinement of control that the elaboration of sound

itself can become a genuine compositional endeavor; this is such a new situation that we are not yet fully prepared to deal with it.

ELECTRONIC AND COMPUTER MUSIC

Since World War II, concrète and electronic music (electroacoustic music) have vastly increased the range of sound material at the disposal of the composer. This has already had a large impact on compositional ideas (even for composers calling for traditional instruments). But electroacoustic music has developed somewhat as a separate branch of music. This separation may be beneficial, even necessary. Bayle contends that the *acousmatic* situation (Schaeffer 1966) brings a large change, just as cinema is not recorded theater, and that this new situation should be dealt with in different ways (Bayle 1977). Yet, significantly enough, some composers who had placed great hopes in these new sonic resources have soon been disillusioned. For instance, Ligeti quit electronic music after producing *Artikulation*, an electronic piece with quite rigorous specifications, because it did not offer him the unbounded possibilities he expected.

If I may simplify the picture here somewhat, I think the limitations of concrète and electronic music can be described in the following way. Musique concrète makes any recorded sound available for musical composition: it thus provides a wide variety of natural sounds with complex structures. But these sounds can be transformed only in ways that are rudimentary in comparison with the richness of the original material; this brings in the danger of capitalizing on sound effects and privileging an aesthetics of collage. Electronic music, on the other hand, allows precise control of the structure of electronic sounds—very simple and rather dull sounds. These simple sounds can be enriched, but only through manipulations that to a large extent ruin the control the composer can exert on them. Of course, these two processes are often intertwined: natural and synthetic sounds can be mixed together, and live electronic music blends instrumental gestures with recorded or electronic sound material. However, the dilemma between richness of sound and refinement of control remains even in these more complex situations.

With computer technology, one may hope to go beyond these limitations. The computer allows precise control over the structure of synthetic sounds. This is not limited to simple tones or noises, as was the case for electronic music before; thus the sounds can be elaborated in complex ways, which allows a composer to enrich them without losing control and reproducibility. With digital techniques, one can now process real sounds as well—like musique concrète does, but also in more subtle and internal ways, through the implementation of sophisticated analysis-synthesis techniques. Loudspeakers, though—

and "hi-fi" in general—have been to some extent designed to accommodate the spectral distribution of orchestral music, and inferior reproduction equipment can dramatically impair the rendering of electronic or computer music with different characteristics (e.g., with a sound structure that fails to mask the distortion or that has energy concentrated in the very low or the very high frequency range). What makes it worse is that the listener of computer music has no internal reference of what is to be heard, contrary to the case of recorded instrumental music.

At any rate, much of the music realized with the computer so far has used the new possibilities in ways that can be considered either rudimentary or derivative. More often than not, composers have merely displaced procedures developed with media (the musical instruments) having completely different constraints. But computer music is a recent field; certainly it is difficult to use the computer in musically creative ways. A substantial amount of research is still necessary to take full advantage of the digital techniques already available. However, access to digital techniques should vastly improve in the coming years with the advent of digital synthesizers.

DIGITAL SYNTHESIS: THE PSYCHOACOUSTIC PROBLEM

Direct digital synthesis, developed by Max Mathews in 1958, involves a computer directly controlling a loudspeaker through a digital-to-analog converter (DAC). Direct digital synthesis by computer is the most general sound synthesis process available.

Mathews developed programs to use this method efficiently. In Music V, the user is provided with building blocks (*unit generators*), which the user may assemble as desired. Such assemblies of unit generators (and the procedures associated with them) are triggered by events (corresponding to instruments "triggered" by notes). When selecting a combination of blocks, the user can configure the program to the desired level of complexity (Mathews et al. 1969). The user can also add new blocks (e.g., unit generators implementing processes like VOSIM [Kaegi and Tempelaars 1978] or waveshaping [Arfib 1979; LeBrun 1979]). Subroutines can also be added to Music V to tailor the program to specific compositional needs. Mathews' superb design has served as a model for similar programs (such as Music 4BF, Music 7, Music 360, and Music 11). (Music V was also adapted for a Honeywell DDP 224 minicomputer by Ruiz and me in 1969.) More recently, digital sound synthesizers have been incorporating aspects of Music V into their designs.

Other approaches exist for sound synthesis programs. Truax's POD (Truax 1977, 1978) does not aim at the same generality. Rather, it attempts to provide certain compositional possibilities interactively. Xenakis (1971) has proposed methods for statistically controlling sound samples, which he claims embodies an altogether different conception, manipulating complexity directly instead of building it up from simple elements. However, it is not clear that these methods can give essentially different results (Smith 1973).

Given a program that implements a number of sound synthesis procedures, the composer must know how to describe the sounds to be generated in terms of these procedures. By contrast, a composer for conventional orchestras knows the sounds of the instruments from long experience and training, and hence has little need to know how they work physically. But direct digital synthesis forcefully raises this problem, which we call the *psychoacoustic* problem: providing an adequate physical description of interesting timbres.

Direct digital synthesis demands a great deal of the computer, which for each second of sound must put out tens of thousands of numbers computed according to the prescribed recipes. Until recently, this has usually not happened in real time. (That is, it takes more than 1 sec for the computer to generate the samples corresponding to 1 sec of sound.) So the physical description of the desired sounds must be provided in advance. The user cannot hear the sound while varying parameters and manipulating knobs, as in electronic music. The musician has to resort to some psychoacoustic knowledge relating the physical parameters of a sound and their perceptual effects.

The first users of direct digital synthesis immediately encountered this fundamental problem. Even familiar sounds, such as those of traditional instruments, are not as easy to imitate as one might think. Early attempts to imitate sounds using descriptions from classical acoustics treatises failed, pointing out the inadequacy of these descriptions and the need for more detailed and relevant data. Also lacking was information about how to impart liveliness, identity, and personality to synthetic sounds. Hence the initial outcome of computer music was somewhat disappointing. The sounds produced were rather dull, and the new possibilities did not seem to live up to the expectations.

COMPUTER STUDIES OF INSTRUMENTAL TIMBRES

These early failures initially cast doubt on the sonic potential of the computer, especially since a number of other composers had been using the computer merely as a more complex and precise sound synthesizer. (However, direct computer synthesis of sound by Max Mathews in 1958 was implemented before the widespread commercial application of analog synthesizers in the late 1960s.) These composers controlled physical parameters of sound like frequency, waveform, and attack and decay times of the amplitude envelope. Hence, they were using sounds very much like those produced by electronic organs, both in their sound structure and their aural effect. This gradually changed, especially when computer studies of instrument tones provided a better understanding of the complex structure of these sounds and the aurally relevant parameters of this structure. The computer has since produced realistic simulations of various instruments, including brass, pianos, and strings (Risset and Mathews 1969; Morrill 1977; Schottstaedt 1977; Risset and Wessel 1982).

Incidentally, the pioneers of electronic music failed to imitate realistically the sounds of traditional instruments. This failure had been ascribed to the inadequacy of Fourier analysis. But the early analyses were not time-varying, so the failure was due rather to the fact that the imitations were derived from an oversimplified model of instrumental sounds. This model comprised a steady state with an invariant spectrum presumed to be characteristic of the instrumental timbre. Some acousticians (Stumpf 1926; Leipp 1971) and tape musicians (Schaeffer 1966) were not as naive in this respect; the inadequacy of such a model could easily be demonstrated by simple manipulations such as tape reversal. However, it was not easy to implement more elaborate synthesis models. Computer studies (through analysis of the real sound and synthesis of various simplified models) demonstrated that, rather than being associated with a given set of physical parameters (e.g., a given spectrum), a given timbre can often be related to some property, some law of variation, some relationship between the spectrum and other parameters. For instance, brass timbres are predominantly characterized by increases of the high-frequency components related to increases in loudness (Risset and Mathews 1969); bowed-string sounds are characterized by a jagged frequency response (fig. 7.1), which causes a characteristic complex spectral modulation with the slow, quasi-periodic frequency modulation known as *vibrato* (Mathews and Kohut 1973). That these features indeed characterize string and brass timbres to a large extent can be demonstrated with Mathews' electronic violin, which can sound like a real violin, but also like a trumpet if it is given the spectral characteristics of the brass family (Risset 1977; Roads 1980).

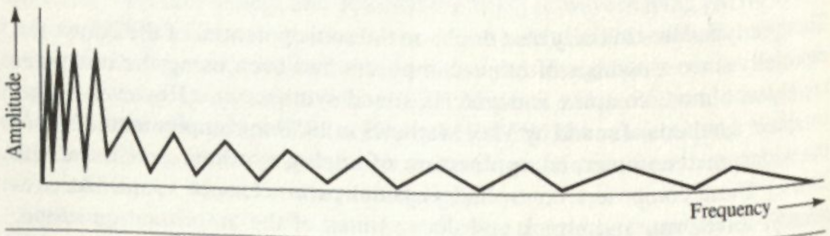


Fig. 7.1. A simplified version of the jagged frequency response of a bowed-string sound

The brasslike timbres synthesized by Morrill (1977) exemplify two points. First, this synthesis has drawn upon the work of several people: Mathews (the author of the synthesis program), me (characterizing brass by a law of spectral variation), Chowning (1973) (who invented the frequency-modulation technique that permitted an elegant implementation of this variation), and Morrill himself (who brought a thorough understanding of trumpet playing to improve the synthesis). Hence the computer facilitates cooperation of researchers, even working years or thousands of miles apart.

Second, Morrill increased the realism by inserting "wrong notes" and "accidents"—which demonstrates that computer synthesis is not necessarily cursed with ice-cold perfection. In fact, listeners can be biased into believing that synthetic sounds are real and that real sounds are synthetic—with more than chance probability, if the synthetic sounds exaggerate a property thought typical of real sounds. This is not simply for the sake of deceit. It may well be that perception has developed sensitive mechanisms that are on the alert for detecting typical accidents that serve as a signature of the origin of the sound. If this is so, the feelings conveyed by sounds devoid of such accidents might be incomplete and lack vividness.

IS INSTRUMENT RESEARCH OR IMITATION WORTHLESS?

At this point I would like to answer an objection that may be in the mind of the reader. It may seem rather conservative to apply a general process like direct digital synthesis to the mere imitation of conventional instruments. Of course the researchers who created these imitations thought of this objection. They were attracted to digital synthesis for its new sound possibilities, and they first tried to synthesize "new" sounds by implementing some preconceived models of these sounds or by varying various physical parameters at random. The results were very disappointing: it quickly turned out that the models were not very fruitful; random variation of parameters was akin to groping and did not lead anywhere. (Random variations in a real-time situation are discussed later.) It became clear that they did not even know which parameters to vary in which domain to make variations really significant to the ear: the psychoacoustic problem was very much there. I studied the trumpet in 1964, not to develop an ersatz, but primarily to try to understand what is aurally relevant—not only what is "pleasant" and "lively," but also what gives the sound a characteristic identity. In short, the long-term goal of instrumental research is not imitation.

However, imitation is not as futile as it might appear. The gamut of sounds available to the composer would not be complete if it did not include sounds similar to those of traditional instruments, which are after all the material for many compositions past and present. There is also considerable interest in mixed works combining live performers and tape sounds, because of the visual interest of such presentations and for good acoustical and musical reasons. Furthermore, computer synthesis of instrumentlike sounds permits the composer to develop subtle relationships between tape and instruments. The synthetic and instrumental sounds can be controlled with comparable refinement, even though the types of control are different. For example, in his piece *Studies for Trumpet and Computer* (1975), D. Morrill used brasslike synthesis to establish a connection with the natural trumpet sound and then extend it beyond its normal range. I used similar processes in pieces like *Dialogues* (1975), where one of the two media (instruments and tape) develops or prolongs sound structures introduced by the other. The instruments and the tape often do not use the same scales, even though the scales are related through harmonic structure. For example, an instrumental chord can be followed by a gonglike sound (see fig. 7.2) whose components are the fundamentals of the chord (Risset 1968). Hence, this sound is not really new in timbre synthesis, but it is new in the sense that it is related to the harmony.

Other striking types of nonconventional control of instrumentlike sounds have been developed as a consequence of instrumental studies. These studies helped Chowning realize the advantage of his frequency-modulation (FM) technique. This remarkable invention provides an elegant way to control the variation of the most salient features of spectra (harmonic or inharmonic), which is essential to creating timbres with both identity and liveliness. Chowning and his collaborators have used this technique for interpolations between different timbres, as in a piece by R. Harvey where pianolike sounds are gradually transformed into chimes. Using instrumental tones, Beauchamp (1975) has derived nonlinear models that may have great potential, especially with the development of nonlinear distortion techniques for spectral synthesis (Arfib 1979; LeBrun 1979). Grey (1975) has also drawn valuable information from instrumental tones. He has been able to vastly reduce the data necessary

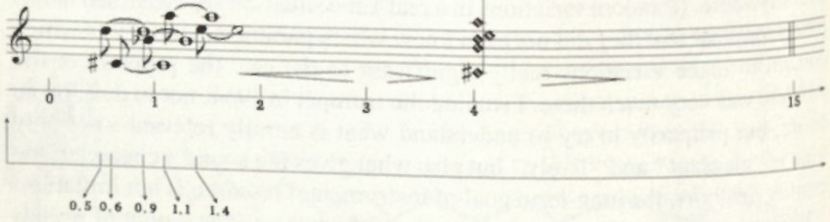


Fig. 7.2. The harmony of the notes played in the first 2 sec echoed by the timbre of the sound played at 4 sec (from *Mutations* on INA-GRM AM-564-09).

to represent them and to isolate characteristic features. He has studied the subjective timbral space for instrument sounds (see also Wessel 1973) and has performed continuous interpolation between instrumental timbres. One can imagine games of timbre mirrors and metamorphoses made possible by timbral interpolation. For example, a piece (1979) realized by G. Bennett at IRCAM associates a singer on stage with synthetic voicelike sounds that can be close replicas or remote, idealized images of the real voice. This striking relationship reflects the poetic theme of the music. Thus, the computer can open up the whole continuum of timbre—among, as well as beyond, the sounds of the voice and the traditional instruments.

To date, most instrumental studies have focused on isolated tones. Despite this limitation, they have gone a long way in clarifying the scene. Yet they are only a beginning: the frontier is now at the level of the musical phrase: What are the "prosodic" variations of the tone parameters? (Isolated tone studies were of course necessary to unravel these parameters. Moreover, the isolated tone situation probably provides the most stringent test [Grey 1975].) Mathews, Morrill and Strawn have begun to tackle the problem of phrasing. Such studies should enhance our understanding of the prosodic features that are style-dependent and the way in which they are style-dependent.

NEW POSSIBILITIES: INHARMONIC TONES AND FUNCTIONAL SPECTRA

Instrumentlike tones can be transformed in subtle ways. For example, it is possible to construct an inharmonic bell-like tone with a more or less distinct pitch in a fashion similar to building up a chord. That is, the amplitudes of selected frequency components all follow the same amplitude envelope (for example, an abrupt attack and a long decay). However, in contrast to a standard chord, the amplitude envelope of each component is given a different duration. Then one can take the same frequency components and apply a different amplitude envelope—one that builds slowly. As a result, the character of the tones changes. Because of the different lengths of the components, they do not swell in synchrony. Hence, instead of fusing into a bell-like sound, the components are diffracted, although the underlying harmony remains the same. Thus, by changing a single function in the score, one can change the internal structure of a sound. (I have used such techniques in my piece *Inharmonique*.)

John Chowning has recently synthesized beautifully realistic sopranolike tones (1980). By applying different sorts of frequency modulation to different voicelike spectra, he can make voicelike sounds emerge from an undifferentiated electronic sound texture. In addition to yielding new expressive possibilities, this work sheds light on the aural analysis of sonic masses; this phenomenon indicates that the ear takes advantage of microtemporal differences to sort out different signals. This effect of micromodulation, originally suggested by Michael McNabb, is confirmed by the studies of McAdams (1982).

Both these examples refer to a vast domain opened up by digital synthesis, namely that of *inharmonic tones*. Most sustained instrumental tones are quasi-periodic, and their frequency components are harmonically related, which stresses certain intervals like the octave or the fifth. With the freedom of constructing tones from arbitrary frequency components, one can break the relationship between consonance-dissonance aspects and fixed, privileged intervals (Pierce 1966). In his piece *Stria* (1977), Chowning has thus been able to make rich textures permeate each other without dissonance or roughness, by controlling the frequencies constituting these textures. This is also a case where spectra not only play a coloristic role (see Roads 1985), but actually perform a *quasi-harmonic function*. Although the idea of articulating musical structure through timbral organization is not new, one needs the control that computer synthesis provides to truly achieve it. Is the aural differentiation of spectra fine and reliable enough to allow rich possibilities in music? As Mathews has remarked, the example of speech indicates that spectral differences can form the basis for a refined code of communication; hence, one may hope that the fine spectral control obtainable through digital synthesis can play a fully functional role in effectively articulating entire compositions (fig. 7.3).

ILLUSORY SPACE

Sounds (either recorded or purely synthetic) can be processed by the computer so as to give the illusion that they are moving rapidly across space. This technique was developed by Chowning (1971), whose goal was to achieve control over the movement of sounds in space. This computer technique for projecting sound in space is both more refined and practical than using arrays of speakers (e.g., up to 800 were used in the Osaka World's Fair in 1970). It is also interesting to note that Chowning was able to achieve the illusion of a vast space in which virtual sound sources were heard as moving, by using purely synthetic cues. This was possible thanks to the precision of the computer; but it also required psychoacoustic understanding of the physical parameters responsible for the subjective evaluation of sound. One can hope that psychoacoustic investigations will go further in enabling us to use the computer to directly control cues for this or that subjective aspect of the sound and to create an illusory auditory world by acting more directly upon the perceptual operations that can evoke certain auditory experiences. Auditory illusions or paradoxes that can be

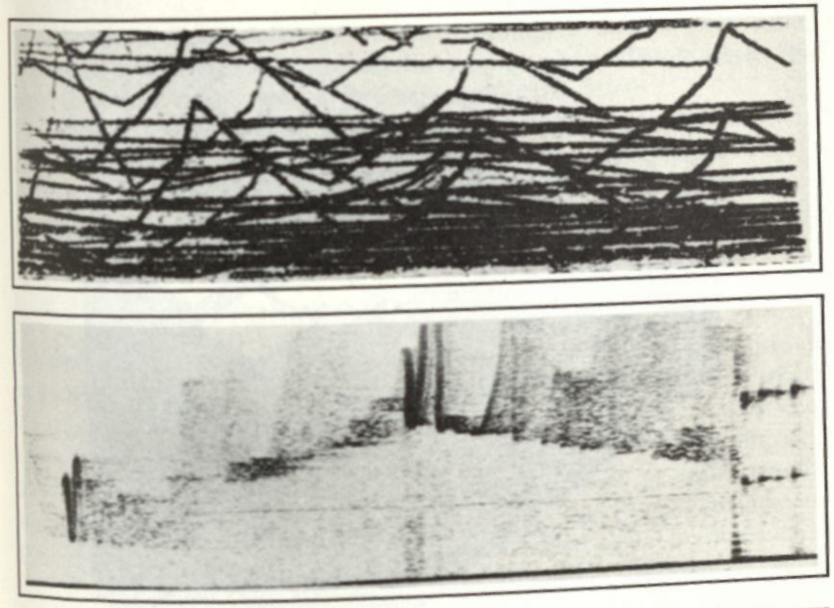


Fig. 7.3. Sound spectrographs of different sound textures obtained by computer synthesis. These are frequency-versus-time plots, with darkness indicating intensity. Each figure displays 1–2 sec of sound. Part (*a*) shows a thick texture of gliding inharmonic tones; (*b*) shows tones with sweeping frequency curves (from *Songes*).

demonstrated with the computer indicate that this is not merely utopian thinking.

PARADOXES OF PITCH AND RHYTHM

Shepard (1964) demonstrated an aural counterpart to the Penrose/Escher illusion (fig. 7.4). He produced a sequence of 12 tones in chromatic succession that seem to rise indefinitely in pitch when repeated. I extended this paradox by synthesizing ever-ascending (or descending) glissandi and sounds going down the scale while getting shriller. Here computer sound synthesis made it possible to precisely arrange physical parameters in ways that are not encountered in nature. Yet these paradoxes are not merely *truquages*—that is, artificial curiosities: they reflect the structure of our pitch judgments. For special cases (e.g., sounds with octave components), pitch splits into two attributes: a *focused* aspect, related to pitch class, and a *distributed* aspect, related to spectrum (hence to timbre). The paradoxes are obtained by independently controlling the physical counterparts of these attributes, which are normally related.

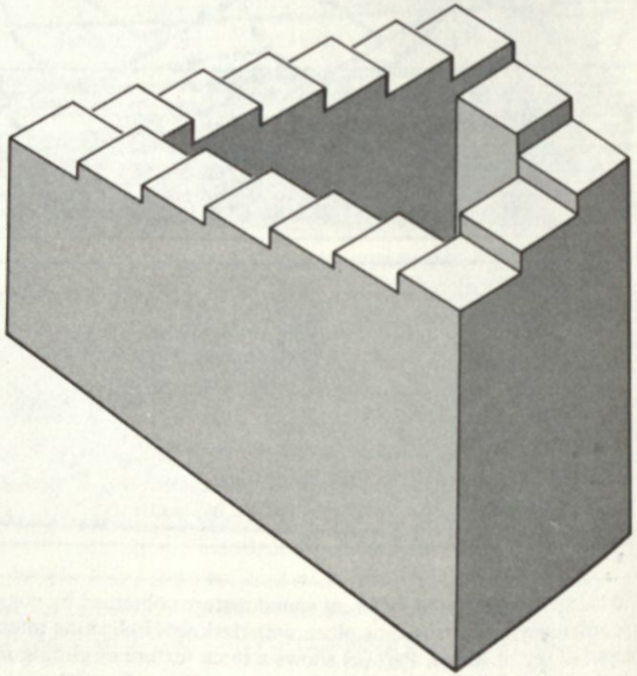


Fig. 7.4. The staircase illusion, originally drawn by Penrose, after which Escher's famous waterfall illusion was modeled

Such a dichotomy of pitch does not appear in every situation. Yet one cannot help relating it to Schoenberg's conception: "It is my opinion that the sound becomes noticeable through its timbre, and one of its dimensions is pitch. . . . Pitch is nothing but timbre measured in one direction" (1922). Schoenberg goes on to the idea of the *Klangfarbenmelodie*—melody based on timbre variation. In the above case one can independently control "tonal" and "spectral" pitch. By divorcing the physical cues for these two aspects, one can make sounds that go down in pitch (for most listeners) when their frequencies are doubled—when one doubles their playback speed (Risset 1977). This shows how misleading mere intuition can be in predicting the effects of simple transformations on unusual sounds.

In 1974, Knowlton synthesized beats that seemed to speed up forever. I extended this paradox and coupled it with the pitch paradox to produce, for instance, a sound that goes up the scale while it is getting lower and speeds up

but at the same time gradually gets slower! Once again, such oddities reflect mechanisms of perception. They reveal the complex nature of pitch and rhythm perception, which can be relevant to musical practice.

In my pieces *Fall* (from *Little Boy*), *Mutations*, and *Moments Newtoniens*, I used such paradoxical sounds. In *Fall*, the endlessly descending glides convey a feeling that, although novel, soon appears familiar. Thus the possibilities of manufacturing new sonic structures may subtly extend our perceptual set (fig. 7.5).

IS PSYCHOACOUSTICS IRRELEVANT?

Some musicians dismiss psychoacoustics as irrelevant for musical purposes. In an interesting discussion, Randall (1967) makes some provoking observations. For instance, he deems it essential for the psychoacousticians of music to keep in touch with current concerns in composition, even to the extent of doing composition themselves—since it requires compositional skill to appreciate the musical relevance of a test and the contexts it implies, and to evaluate the compositional possibilities of sound material. This is indeed a vital point. Much of the work I have mentioned was developed by composers. Yet they did act as psychoacoustic researchers—asking the right questions with the help of their musical competence, but trying to solve them through genuine experimentation. In doing so, they put their own compositional set and even their aesthetic viewpoint into the background. (This is not meant to question their sense of artistic commitment as musical creators; after all, research is a different stage of creative work.) These examples indicate that this attitude was valuable in giving access to new domains as well as in providing a deeper understanding and control of familiar domains.

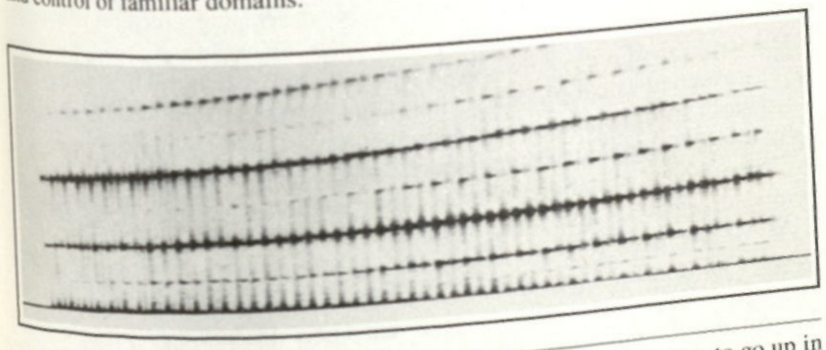


Fig. 7.5. Sound spectrograms of a study of paradoxical sounds that seem to go up in pitch but eventually reach a lower pitch. They also seem to speed up but eventually become slower (from *Moments Newtoniens*, on INA-GRM AM-564-09).

However, I cannot fully agree with Randall when he condemns the use of "contextually non-significant ingredients chosen to instrumentalize, i.e. lush up, electronically generated timbres" (p. 135). One can argue that it is the professional responsibility of the composer, when using direct digital synthesis, to properly lush up tones, since no instrument-maker or performer will take charge of that. Then Randall claims:

> The often-deplored uniformity, monotony or outright nastiness of electronic timbres seems to me more properly analyzed as a failure of some existing electronic compositions adequately to structure and develop their timbral components as elements of the composition, rather than as any inherent debility in current technology or any musical dullness "inherent" even in the balder electronic timbres (pp. 135–6).

I answer that this seducing statement will convince me only after I have heard a composition using simplistic, dull, "electronic," or "computerish" sounds that does not alienate demanding listeners. (Among such listeners, Edgard Varèse was especially demanding.)

But, of course, compositional ambition should not stay at this superficial level of tone congruence. In any case, the composer using computer synthesis, even if contemptuous of psychoacoustics, is likely to use some psychoacoustics—but in an implicit and fragmentary way, which may hinder a full realization of the potential of the new sonic material. Randall questions the existence of psychoacoustical facts about the material of music. Indeed, psychoacoustic data can never fully determine what happens or should happen in the area of music; however, they can provide valuable guidelines for developing fruitful and consistent musical logic in new sonic domains. When Randall dismisses questions about timbre and favors "developing structures of vibrato, tremolo, spectral transformation," and other dimensions of sound, he in fact refers to physical properties. As Wessel indicates (in an unpublished essay), he gives to such physical properties the status of musical dimensions (in that they form the basis of compositional manipulation), which is taking a lot for granted as to the perceptual structure of such properties. This attitude is a heritage of the instrumental situation, where a closer relationship exists between what the composer specifies and what the listener hears.

As Varèse often remarked, genuine success in architecture results from a genuine understanding of the building material, not from imposing structures developed with other materials. To invent a new structure appropriate to solve the problem at hand, the architect cannot exert creative genius *in abstracto*, but must thoroughly investigate the properties of the new material. Similarly, the new sound material portends new compositional possibilities. For instance (as

demonstrated previously), tones made up of nonharmonic partials may inspire different melodic and harmonic constructions and revive the concept of scale (cf. Pierce 1966). However, this requires understanding of the conditions under which inharmonic components fuse into a single tone—a psychoacoustic problem. As was advocated in the introduction, much psychoacoustic exploration is required to suggest new musical architectures and to test their viability.

Sounds propagate across distances and around obstacles, and hearing is well equipped to be on alert: it is mostly sensitive to changes, and it tends to turn off or forget stable and steady sounds. It is not surprising that hearing does not rely only on the exact structure of the spectrum to evaluate timbre and identify the origin of the sound, since this structure is often severely distorted during propagation through the air. Hearing is very sensitive to frequency aspects, which are only rarely modified between the sound source and the listener. (A notable exception is the Doppler effect—but then hearing extracts from it information on the source of movement.) Hearing is, on the contrary, quite insensitive to the phase relations between the components of a complex periodic sound, which is fortunate because these relations are smeared in a reverberant environment. Timbre is related to rather elaborate spectral and/or temporal patterns resisting distortion; from these elaborate patterns, hearing has intricate ways to also extract information about loudness and distance.

What are the musical implications of these environmental considerations? In music, hearing works in a gratuitous way, and so do the functions of alarm detection or sound source recognition. If the complex mechanisms responsible for these functions are not invoked, it may well be that something will be fundamentally missing in the perception. This point has already been stressed with respect to "accidents" adding realism to instrumental sounds.

To avoid a dull, "electronic" quality in computer music synthesis, one must prescribe constant changes in the physical parameters—changes from tone to tone as well as throughout each tone. (See example 100 of my *Sound Catalogue* [Risset 1968]). This is again confirmed by Chowning's vocal synthesis, where both random and systematic variations are programmed. (This has implications for desirable software for computer music systems; it would be helpful to facilitate such a *synthesis by rule*.)

Many electronic and computer pieces (using simple steady tones) achieve no real contrast in loudness, even though there may be strong differences in the physical intensity of the sounds. The more complex loudness-detection mechanisms rely on spectral variations as well as distance cues, so that whispered speech or loud trumpets are perceived as such regardless of distance, even if the sound level of the former is higher. But in many synthesized compositions, these loudness detection mechanisms are not allowed to come into play, and the effect is very flat. As a counterexample, my *Mirages* attempts in simple ways to suggest cues for sound paths in an imaginary world.

The illusory space conjured up by Chowning (1971) does not simply recreate a familiar environment or lush up tones through reverberation. It genuinely adds depth (in the figurative sense) to sonic perception. Highly developed and intricate perceptual operations probably detect clues about the origin and the localization of the sound. Such clues guide the mind into constructing an inner representation of the world consistent with the aural data. A thorough understanding of these operations might enable us to activate them through artificial sounds. This should give to those sounds pregnancy, presence, and identity deeply anchored in perception, through the fulfillment of elaborate perceptual operations. If this is not entirely utopian (as I said, I believe the loudness and depth examples indicate it is not), clearly much more is at stake in psychoacoustic investigation than just lushing up tones!

HIGHER-LEVEL STRUCTURES

It is difficult to draw a dividing line between composition *of* sounds and composition *with* sounds. Composing a sound may in fact consist of developing a process at the sonic level. As we develop our knowledge about music and circumscribe our own musical desires, computer programs can help us develop processes that we judge musically useful. Of course, this is a dialectic process, since any language for the description of sounds encourages certain types of manipulation and transformation and suggests trying these first. Music V itself, although in a relatively minimal way, facilitates certain types of sound processing. Score (Smith 1972) goes further in supplementing, among other facilities, interesting ways of manipulating motives (repetitions, transpositions, inversions, etc.). POD (Truax 1977) is a composition/synthesis system; the user can control the range in which the program will randomly select sonic parameters. Programs like SKETCHPAD (Wessel 1979) aid the composer in exploring selected sound material. Musical input languages like Pla (Schottstaedt 1983) and Formes (Rodet and Cointe 1984) afford convenient and powerful possibilities.

Artificial intelligence researchers have exerted their considerable ingenuity to devise advanced formal languages. These languages can make it easier for the user to specify certain types of musical processes. Such developments may bring an extension of the structural role of musical notation. (I prefer the French term *écriture*, which has wider implications.) Thanks to notation, music can be replayed, studied, and corrected; but notation has also been important in the historical evolution of Western music in that the latter has favored specific types of transformations that are not "natural" if one stays in

the realm of sound. This would include spatial symmetries, which developed to a high degree of elaboration in the music of Machaut and Ockeghem. Resorting to the computer implies describing processes in some formal language, which is a kind of notation.

Advanced formal languages could favor highly specific types of sonic and musical transformation. Yet the "flesh of perceptual organization" (Arnheim 1968) should not stay unmoved by these transformations, which should touch upon sensitive areas. Hence they should be studied in close relation with the investigation of the sonic material. In this manner, specific types of musical manipulations could be explored; also, new concepts of data organization could suggest convenient ways to present the complex data required for the synthesis of sound. Geometric models providing useful representations of sound structures and of their transformations should help in organizing musical thought. Proper graphic capabilities could prove invaluable in that respect. They should bring deeper understanding to the relationship between the aural and the visual domains, which is in itself a problem of considerable interest. So one can look forward to the mutual reinforcement of advanced music languages and thorough exploration of the potential of the sonic material. I hope this will provide powerful conceptual and practical tools. One may be tempted to use advanced, "intelligent" languages to their fullest extent. However, I believe that the creative process, to a large extent, resides in setting new conditions and inventing new rules, so I expect that utter formalization and automation might offer only academic interest (and results). As Debussy said, works of art make rules but rules do not make works of art.

PROGRESS AND DIFFICULTIES IN DIGITAL SYNTHESIS

The previous account has emphasized the considerable progress made in digital synthesis. Moreover, results of synthesis and psychoacoustic research are disseminated in the form of articles, catalogs, and lexicons (Chowning 1971; Morrill 1977; Moorer, Grey, and Snell 1977; Moorer, Grey, and Strawn 1977, 1978; Schottstaedt 1977; Lorrain 1980; Haynes 1980). To distribute some results of timbre exploration, I assembled a catalog of computer-synthesized sounds (1968). I gathered both the recordings of the sounds (so listeners could evaluate the timbres) and the listing of the synthesis data (which included a complete and precise description of the sounds and the recipe to resynthesize them). The sounds included in the catalog, presented as examples, were by no means models. They were intended to be starting points for systematic explorations, rather than a library of ready-made sounds. In this way, digital synthesis with programs like Music V facilitate the exchange of information. Psychoacoustic expertise can accumulate cooperatively to increase the gamut of sounds available through digital synthesis.

Yet, despite its advantages in terms of power, reproducibility, and communicability, direct digital synthesis still appears to many composers as difficult, almost forbidding. In their impatience, they often initially produce sounds that are not elaborated in a refined way, which they find unsatisfactory. As discussed earlier, the problem of obtaining rich and supple sounds is a delicate one indeed—and the delay between the specification and audition of the sound does not make it easier.

Many musicians think that the remedy to their difficulties in obtaining adequate sounds with the computer lies in one of two processes that are being explored at the moment, namely real-time interaction and processing of natural sounds. Although I am very interested in these processes, I would like to draw attention to the problems they generate. I think there are overexpectations concerning their ability to solve musical problems involving timbre.

REAL-TIME POSSIBILITIES

Real-time computer systems could be developed effectively only with the advent of minicomputers. The first such systems were hybrid: the computer provided control signals for analog sound synthesis equipment (Kobrin 1977; Oppenheim 1978). Thus it was relieved of the burden of computing all the details of the waveform. Genuine real-time systems allow modification of the sound while it is being produced, which is invaluable, since one can take advantage of aural feedback to react and to introduce performance nuances. However, total real-time operation puts great demands on the user, who must perform or improvise in an instrumental fashion—or resort to automation. (In contrast to the heavy demand on the capabilities of general-purpose computers in synthesizing sound, the problem is to impose meaningful control over the flow of the sound going by. Interest in cascading sequences and in controlling random excursions of parameters may be ephemeral.)

In the design of GROOVE, a real-time hybrid system, Mathews and Moore (1970) gave due consideration to the question of which specifications should—and should not—be made in real time. In particular, the system enabled the user to exert control over the sound analogous to that which a conductor can exert over an orchestra. Clearly, such a system leaves a lot of room for the influence of the performer. In fact the gestures can control compositional processes as well as performance nuances. The contributions of composition and performance can either stay distinct or merge intimately.

Of course, analog equipment is not as precise or flexible as digital equipment. Fortunately, the progress of technology has permitted the development of a number of purely digital real-time sound synthesizers.

The development of these digital synthesizers is extremely promising. They help bring together the generality of digital sound synthesis with appealing real-time possibilities. Until recently, it was possible to use the computer for music only in large institutions. But powerful music systems may become available at a low price, and this new economic situation will completely change the status of digital electronic music. Digital systems are becoming private tools for the independent composer.

In the long range, this expansion of the digital techniques in music will probably have far-reaching consequences beyond the professional music scene. Mathews' experiments indicate that it is possible to develop digital systems that can be used in a variety of ways ranging from a "record player" situation (where the "performer" has little control) to an instrumental situation (where the performer has total control but is also submitted to considerable demands) (Roads 1980). Between these two extremes one may have many different types of control—like the situation of the conductor, who does not produce all the notes, but who "performs" significant control. Such a system could offer genuine musical responsibilities to the user without necessarily demanding the technical prowess of a professional musician. This is utopia, but making such systems available to the public (which is already economically conceivable) might restore contemporary musical practice. It would help fill the gap between amateur instrumentalists and this music which they presently do not relate to their musical practice. In this utopia, professional musicians could make proposals of pieces to be played as such or to be completed or assembled in a variety of ways, and there would be a continuous gamut of degrees of initiative or responsibility that the listener-performer could take. Even if such systems do not become widespread among the public, they should offer new and challenging situations to musicians. Needless to say, the design and implementation of such systems will take considerable ingenuity and know-how—dependent upon research in electronics, psychoacoustics, and music.

Turning back, however, to the problems of timbre, one should be warned against hasty optimism. With properly equipped digital synthesizers, one can indeed "tune" timbres by trial and error, manipulating knobs with aural control. But the composer should not think that one can generate a huge variety of timbres with the help of these knobs and an infallible musical instinct. What is available to the real-time user is not the whole of the machine's resources, but only those resources that have been set up in advance of real-time operation. As with the Music V program, the synthesizer must be configured to manufacture the sound according to selected models—and it is generally more difficult to reset this configuration for a digital synthesizer than for a program like Music V. Also, improvisatory procedures tend to capitalize on established processes rather than newly found ones.

Although real-time synthesizers have been around for a few years, most of the important and recent work in musical psychoacoustics has been achieved with non-real-time systems. Here again, groping at random (unless one is within some specific framework) may not lead anywhere (Pousseur 1970). Pierce remarked that by randomly varying the weight of, say, 10 frequency bands covering the audible frequency range, one is unlikely to get close to a flat response. And recently it was found by experience that one could not succeed in synthesizing speech properly by adjusting the parameters of a synthesizer in real time; it was necessary to enter speech synthesis data. Real-time operation thus does not permit one to dispense with psychoacoustic know-how, guidelines, and directions. Moreover, the real-time constraint imposes a limitation—which can be severe—on the complexity of the sounds that can be produced. Thus, composers using a digital synthesizer often resort to layering techniques with a multitrack tape recorder—which of course hinders real-time operation.

I am playing the devil's advocate here. Clearly real-time operation has a lot to offer, but it can also be a mixed blessing. Manufacturing sounds through empirical manipulation will not by itself provide the musician with adequate sounds, unless it is supported by sonological and compositional research.

PROCESSING OF NATURAL SOUNDS

Another way to obtain rich sounds is to take richness where it lies—that is, in natural (e.g., instrumental) sounds. In other words, digital systems can be used for processing natural sounds rather than for performing sound synthesis. This is indeed an interesting direction. With analog-to-digital converters, sounds can be converted into numbers. These numbers can be stored and later converted back into sound, but they can also be digitally processed. (Digital recording provides higher-quality reproduction and better long-term storage than analog recording techniques.) For instance, digital mixing by computer is a demanding but very precise process. In my piece *Mirages* (1978), I used digital mixing to contrast a chamber orchestra on stage with an imaginary orchestra on tape. The imaginary orchestra consisted of a polyphonic texture obtained by digitally mixing only five short motives recorded separately by ten instrumentalists. In several instances, the sonic material was ordered with the help of timbre maps provided by the SKETCHPAD program (Wessel 1979). The mixing program was the IRCAM version of Music V, which has been

augmented by Gardner and Richer to enable it to accept digitized sounds so that sounds of any origin can be processed by Music V modules. Most of the transformations that can be performed with analog means can also be performed with the computer (often with less convenience, but with more precision and reproducibility). Such techniques were used extensively in the piece *Arcus* by York Höller, realized with the aid of Stanley Haynes at IRCAM.

I mentioned previously the limitations of analog processing: transformations are rudimentary by comparison to the possible richness of the original material, so they often remain limited and superficial. For instance, if one wants to change the frequency of a tone in a simple way, either in the analog or in the digital domain, the duration is also changed accordingly, whether one likes it or not. To perform more refined transformations—for example, to divorce changes in frequency and duration—one must somehow go through a process of analysis that gives access to the inner parameters of the sound, modify these parameters, and then resynthesize the sound.

An instance of such a process is *linear predictive coding*, often used for speech (Cann 1979–1980). Moorer (1979) has studied predictive coding at Stanford and at IRCAM. Resynthesis shows little difference from a digital recording of the original voice. However, one can modify the inner parameters available from the analysis. This permits extensive changes in the speed of the voice without altering the pitch or vice versa. This is possible because the analysis permits separation of the parameters corresponding to the glottis (which determines the pitch) from those describing the vocal tract (which determines the articulation). From this analysis, one could also, for instance, resynthesize a whispered voice. [See the article by Charles Dodge elsewhere in this volume—Editor.]

Analysis-synthesis techniques can be used in other ways. In his piece *Interphone*, M. Decoust used a poem—actually a recording of the poem—as the kernel of the music. Using similar techniques with the help of Arfib, he analyzed certain sentences to get the amplitude and pitch curves. These curves were then used to control parameters (e.g., amplitude and pitch) of synthetic sounds. One can thus control the synthetic sounds in a supple or "gestural" fashion.

Processing natural sounds is an appealing domain. One must be aware, though, that these processes of analysis and synthesis, essential for intimately transforming digitized sounds, are complicated and demanding. Mastering them demands research and expertise, and they do not lend themselves easily to real-time operation, for practical and sometimes theoretical reasons. So I believe there are presently overexpectations of what real-time digital processing of sounds can bring in the immediate future.

A PLEA FOR "SOFTWARE" SYNTHESIS

The potential of digital sound synthesis is still largely untapped. It is indeed a challenge to synthesize sounds with the desirable richness and identity, but examples have shown that this challenge can be met. Controlling synthesis parameters gives the user complete flexibility in structuring, sculpting, and composing inner aspects of the sound—much more than processing natural sounds (fig. 7.6). Control over synthetic sound does not have to be in terms of numbers; it can be graphical or gestural as well. Cadoz and his collaborators (1978) have conducted basic research on the relationship of gestures to aural feedback and to mechanical feedback under program control. Their work may suggest new processes in this direction.

Such control can be available in real time, although I doubt that this issue is as essential as many people now think. (If one thinks of composition as the creation of new forms acoustically immersed in time, it implies thoughtful activity—elaboration and reflection—which cannot take place under the pressures of real-time demands. Composition also involves synoptic vision, global conceptions, and a spontaneity that also defies real-time constraints.)

I speak of "software" synthesis to stress the importance of keeping systems open-ended, maintaining flexibility to adapt their possibilities to musical needs when they arise. This is vastly preferable to fixing the design and limits of digital music systems on the basis of technical decisions. My position may appear to be a regressive stand, but I am by no means proposing the dismissal of recent technical progress in digital music systems. (For example, Peter Samson's [1980, 1985] digital synthesizer at Stanford University speeds up software synthesis.) I do ask that this progress be considered in a sober perspective and exploited in ways that seem conducive to radically new musical possibilities.

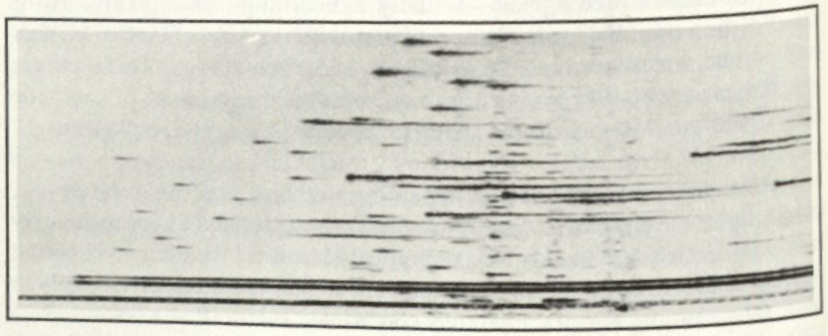


Fig. 7.6. Sound spectrogram of a sonic texture playing on the harmonics of a chord (from *Dialogues*, on INA-GRM AM-564-09).

Such new possibilities clearly raise basic problems with compositional implications. The computer gives access to the continuum of sound and provides a ductile and seemingly neutral material. Does continuum imply indifferentiation? Are discrete categories, quantizing processes, scales, and familiar prototypes essential to nonvague perception and hence to a music capable of stimulating cognition? Is sonic space homogeneous and unbounded—or will perception reveal sharp edges, preferred dimensions, and deeply anchored constraints? And to what extent can (should) composition with computer sounds escape the mainstream of the historic music languages, the "collective norms of musical expression," as put by Amy (1964)? Light on these issues will be shed by the activity of those musicians who elect to work at this new frontier of the development of sound structure.

ACKNOWLEDGMENT

I wish to thank Curtis Roads for greatly improving my English text.

BIBLIOGRAPHY

Alles, H. G. 1977. A portable digital sound synthesis system. *Computer Music Journal* 1(4):5–6. Revised and updated version in Curtis Roads and John Strawn, eds. *Foundations of computer music.* 1985. Cambridge: MIT Press.

Alonso, Sydney, Jon Appleton, and Cameron Jones. 1976. A special-purpose digital system for musical instruction, composition, and performance. *Computers and the Humanities* 10:209–15.

Amy, G. 1964. A propos du "son" et de la "musique." *Mercure de France* March:466–72.

Arfib, D. 1979. Digital synthesis of complex spectra by means of multiplication of nonlinear distorted sine waves. *Journal of the Audio Engineering Society* 27(10):757–68.

Arnheim, R. 1968. Review of *Information theory and aesthetic perception* by A. Moles. *Journal of Aesthetics and Art Criticism* 26(4):552–54.

Babbitt, M. 1964. An introduction to the RCA synthesizer. *Journal of Music Theory* 8:251.

Bayle, F. 1977. Support/escape. *Cahiers Recherche/Musique* 5:13–39.

Beauchamp, J. 1975. Analysis and synthesis of cornet tones using nonlinear interharmonics relationships. *Journal of the Audio Engineering Society* 23:778–95.

Boulez, P. 1971. *Boulez on music today.* S. Bradshaw and R. R. Bennett, trans. Cambridge: Harvard University Press.

Buxton, W. 1977. A composer's introduction to computer music. *Interface.* 6:57–72.

Cadoz, C., and J. Florens. 1978. Fondements d'une démarche de recherche en informatique-musique. *Revue d'Acoustique* 45:86–101.

Cann, Richard. 1979–80. An analysis/synthesis tutorial. Parts 1–3. *Computer Music Journal* 3(3):6–11; 3(4):9–13; 4(1):36–42. Reprinted in Curtis Roads and John Strawn, eds. *Foundations of computer music.* 1985. Cambridge: MIT Press.

Chowning, John M. 1971. The simulation of moving sound sources. *Journal of the Audio Engineering Society* 19:2–6. Reprinted in *Computer Music Journal* 1(3):48–52, 1977.

Chowning, John M. 1973. The synthesis of complex audio spectra by means of frequency modulation. *Journal of the Audio Engineering Society* 21(7):526–34. Reprinted in Curtis Roads and John Strawn, eds. *Foundations of computer music.* 1985. Cambridge: MIT Press.

Chowning, John M. 1980. The synthesis of the singing voice. In J. Sundberg and E. Jansson, eds. *Sound generation in winds, strings, computers,* 4–14. Stockholm: Royal Swedish Academy of Music.

Erickson, R. 1975. *Sound structure in music.* Berkeley: University of California Press.

Grey, John M. 1975. An exploration of musical timbre. Ph.D. diss., Department of Psychology, Stanford University. Department of Music Report STAN-M-2.

Haynes, Stanley. 1980. The musician-machine interface in digital sound synthesis. *Computer Music Journal* 4(4):23–44.

Kaegi, W. 1973. A minimum description of the linguistic sign repertoire. *Interface* 2:141–56.

Kaegi, W., and S. Tempelaars. 1978. VOSIM—A new sound synthesis system. *Journal of the Audio Engineering Society* 26(6):418–24.

Kobrin, E. 1977. *Kobrin: Computer in performance*. Ramona, Calif.: Lingua Press.

LeBrun, Marc. 1979. Digital waveshaping synthesis. *Journal of the Audio Engineering Society* 24(4):250–66.

Leipp, E. 1971. *Acoustique et musique.* Paris: Masson.

Lorrain, D. 1980. Analyse de la bande d'*Inharmonique.* Paris: IRCAM Report 26.

McAdams, Stephen. 1982. Spectral fusion and the creation of auditory images. In Manfred Clynes, ed. *Music, mind, and brain: The neuropsychology of music,* 279–98. New York: Plenum Press.

Mathews, Max V. 1963. The digital computer as a musical instrument. *Science* 142:553–57.

Mathews, Max V., and J. Kohout. 1973. Electronic simulation of violin resonances. *Journal of the Acoustical Society of America* 53:1620–26.

Mathews, Max V., with Joan E. Miller, F. Richard Moore, John R. Pierce, and Jean-Claude Risset. 1969. *The technology of computer music.* Cambridge: MIT Press.

Mathews, Max V., and F. Richard Moore. 1970. GROOVE—A program to compose, store, and edit functions of time. *Communications of the Association for Computing Machinery* 13(12):715–21.

Mathews, Max V., F. Richard Moore, and Jean-Claude Risset. 1974. Technology and future music. *Science* 183:263–68.

Moorer, James A. 1977. Signal processing aspects of computer music—A survey. *Proceedings of the IEEE* 65(8):1108–37. Reprinted in John Strawn, ed. *Digital audio signal processing: An anthology*. 1985. Los Altos, Calif.: Kaufmann.

Moorer, James A. 1979. The use of linear prediction of speech in computer music applications. *Journal of the Audio Engineering Society* 27(3):134–40.

Moorer, James A. 1981. Synthesizers I have known and loved. *Computer Music Journal* 5(1):4–12.

Moorer, James A., John M. Grey, and John Snell. 1977. Lexicon of analyzed tones. Part 1: A violin tone. *Computer Music Journal* 1(2):39–45.

Moorer, James A., John M. Grey, and John Strawn. 1977. Lexicon of analyzed tones. Part 2: Clarinet and oboe tones. *Computer Music Journal* 1(3):12–29.

Moorer, James A., John M. Grey, and John Strawn. 1978. Lexicon of analyzed tones. Part 3: The trumpet. *Computer Music Journal* 2(2):23–31.

Morrill, D. 1977. Trumpet algorithms for computer composition. *Computer Music Journal* 1(1):46–52. Reprinted in Curtis Roads and John Strawn, eds. *Foundations of computer music.* 1985. Cambridge: MIT Press.

Olson H., and H. Belar. 1955. Electronic sound synthesizer. *Journal of the Acoustical Society of America* 26:595–612.

Oppenheim, Dave. 1978. Microcomputer to synthesizer interface for a low cost system. *Computer Music Journal* 2(1):6–11.

Pierce, J. R. 1966. Attaining consonance in arbitrary scales. *Journal of the Acoustical Society of America* 40:249.

Pousseur, H. 1970. Fragments theoriques I sur la musique expérimentale. Brussels: L'Institut de Sociologie de l'Université Libre de Bruxelles.

Randall, J. K. 1967. Three lectures to scientists. *Perspectives of New Music* 5:124–40.

Risset, Jean-Claude. 1968. *An introductory catalogue of computer synthesized sounds.* Murray Hill, N. J.: Bell Telephone Laboratories.

Risset, Jean-Claude. 1977. Paradoxes de hauteur: Le concept de hauteur sonore n'est pas le même pour tout le monde. Seventh International Congress on Acoustics, Budapest. Paris: IRCAM Report 10/78.

Risset, Jean-Claude. 1977. Hauteur et timbre. *Revue d'Acoustique* 42:263–68.

Risset, Jean-Claude, and Max V. Mathews. 1969. Analysis of musical instrument tones. *Physics Today* 22(2):23–40.

Risset, Jean-Claude, and David Wessel. 1982. Exploration of timbre by analysis and synthesis. In D. Deutsch, ed. *The psychology of music.* 1982. New York: Academic Press.

Roads, Curtis. 1980. Interview with Max Mathews. *Computer Music Journal* 4(4):15–22.

Roads, Curtis. 1985. John Chowning on composition. In Curtis Roads, ed. *Composers and the computer.* 1985. Los Altos, Calif.: Kaufmann.

Rodet, X., and P. Cointe. 1984. FORMES: Composition and scheduling of processes. *Computer Music Journal* 8(3):32–50.

Samson, Peter R. 1980. A general-purpose synthesizer. *Journal of the Audio Engineering Society* 28(3):106–13.

Samson, Peter R. 1985. Architectural issues in the design of the Systems Concepts Digital Synthesizer. In John Strawn, ed. *Digital audio engineering: An anthology.* 1985. Los Altos, Calif.: Kaufmann.

Schaeffer, P. 1966. *Traité des objets musicaux.* Paris: Editions du Seuil.

Schoenberg, Arnold. 1922. *Harmonielehre.* Vienna: Universal. Trans. as *Theory of harmony* by Roy E. Carter. 1978. Berkeley: University of California Press.

Schottstaedt, Bill. 1977. The simulation of natural instrument tones using frequency modulation with a complex modulating wave. *Computer Music Journal* 1(4):46–50. Reprinted in Curtis Roads and John Strawn, eds. *Foundations of computer music.* 1985. Cambridge: MIT Press.

Schottstaedt, Bill. 1983. Pla: A composer's idea of a language. *Computer Music Journal* 7(1):11–20.

Shepard, R. 1964. Circularity in judgements of relative pitch. *Journal of the Acoustical Society of America* 36:2346–53.

Slawson, A. Wayne. 1968. Vowel quality and musical timbre as functions of spectrum envelope and fundamental frequency. *Journal of the Acoustical Society of America* 43(1):87–101.

Smith, L. 1972. SCORE: A musician's approach to computer music. *Journal of the Audio Engineering Society* 20:7–14.

Smith, S. 1973. Letter to the editor. *Perspectives of New Music* Fall 1972/Spring 1973:269–77.

Stumpf, C. 1926. *Die sprachlaute.* Berlin: Springer.

Tenney, J. 1969. Computer music experiments. *Electronic Music Reports* 1:23–60

Truax, Barry. 1977. The POD system of interactive composition programs. *Computer Music Journal* 1(3):30–39.

Truax, Barry. 1978. Computer music composition: The polyphonic POD system. *IEEE Computer* 8:40–58

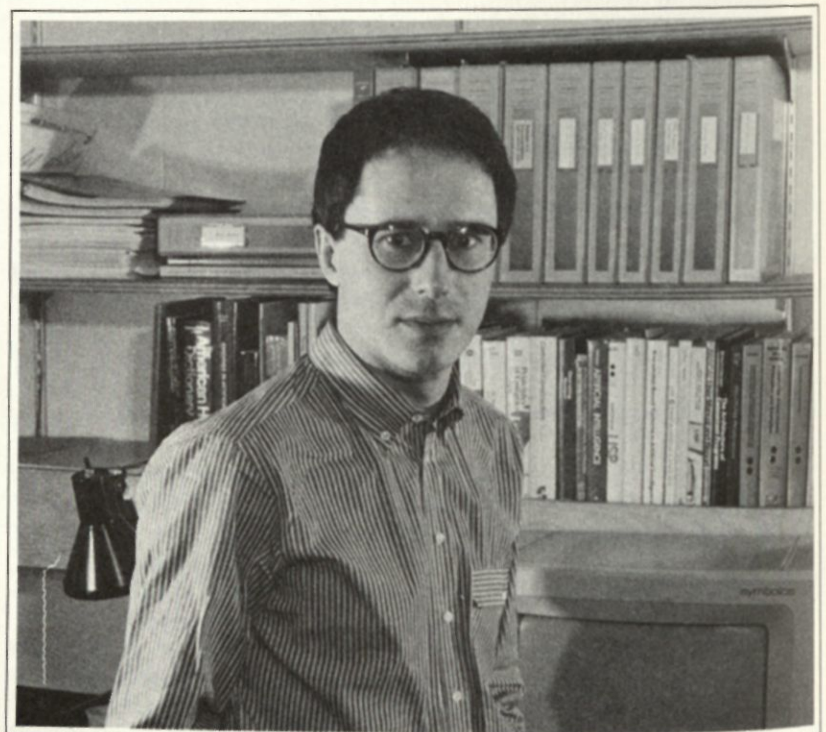
Unesco. 1970. *Music and technology.* Paris: La Revue Musicale.

Varèse, E. 1966. The liberation of sound. In B. Boretz and E. Cone, eds. 1971. *Perspectives on American Composers,* 25–33. New York: Norton.

Wessel, David. 1973. Psychoacoustics and music: A report from Michigan State. *Page* 30. n. p.

Wessel, David. 1979. Timbre space as a musical control structure. *Computer Music Journal* 3(2):45–52. Reprinted in Curtis Roads and John Strawn, eds. *Foundations of computer music.* 1985. Cambridge: MIT Press.

Xenakis, I. 1971. *Formalized music.* Bloomington: Indiana University Press.



CURTIS ROADS

8

THE REALIZATION OF *nscor*

Curtis Roads

This article documents the labor and the technology[1] involved in the creation of *nscor*—a computer music composition recorded on tape. The documentation takes the form of traces left over a long period of work in different locations. No full score to *nscor* exists. Partial scores and notes survive, but because of the extensive mixing, editing, and processing techniques used, they tell only part of the story.

nscor is the third piece in a cycle of four related works, composed over the period 1975 to 1981.

- *Construction* (1975–76)
- *Objet* (1977)
- *nscor* (1980)
- *Field* (1981)

Construction used purely analog electronic sound material; in *Objet*, *nscor*, and *Field*, both analog and digital sounds were mixed. (In any case, all the pieces used processing that obscured the original source of the sounds.) All four compositions were formed out of individually generated sound objects that were mixed and edited into polyphonic textures on a multichannel tape recorder. In all the works, the polyphonic textures were ultimately remixed to a two-channel broadcast and concert version.

Compositional decision-making took place in the presence of sound. Decisions were taken intuitively, rather than being guided by a predetermined formal system. Much listening, selecting, tuning, mixing, and editing went on that could not have been planned away from the studio environment. This is not to imply that I consider more formal approaches to composition less valid. Indeed, I have used such techniques in other pieces.

1. See Appendix A for a list of the equipment used to create *nscor*.

THEORY

The composition theory behind *nscor* emerged from the mode of production. This involved a search for special sounds or short sequences. These sounds were "tuned" to a desired state and then stored for future use.

Since the inception of my work in electronic music studios in 1970, I have gathered sounds in this manner. An advantage to working this way is that one is free (for a time) from the constraint of having to fit each sound into a preexisting framework. The organizational work comes later, influenced by the sounds generated during the initial, exploratory stage.

This approach was influenced by the conditions that prevail in computer music studios. Although composers work with sophisticated equipment and techniques of sound production, they also cope with a paucity of music theory that encompasses the practices made possible by the technology. Specifically, in a time of freshly expanded sound resources, little compositional theory deals with the gamut of sounds and musical processes obtainable in the modern synthesis laboratories. Traditional music theory offers only partial help, since its tenets do not extend very far into music that is based on a broad palette of sound.

Several studies on the synthesis and microstructure of sounds have appeared. The texts by Schaeffer (1966) and Grey (1975) are exhaustive studies of musical tones in isolation. Jean-Claude Risset's *Catalogue* (1968) is a tutorial on the synthesis of many interesting and useful sounds. Even these studies only begin to explore this largely uncharted musical territory.

Sound Objects

One of the more compelling musical concepts introduced in this century is the notion of *sound objects* as a substitute for the traditional note concept.[2] A sound object is a unit of composition in a musical universe where "timbrally complex" (in the absence of a better term) and mutational sound events can exist. A fundamental limitation on the concept of a note is that it is a *single-event* abstraction. Thus it does not well describe the complex entities and sound processes possible in computer music. The static, single-event connotations of the note concept make it difficult to apply to what Varèse termed *sound masses*—sounds that are made up of thousands of smaller acoustic events treated as a compositional unit, such as clouds of granular sounds (Roads 1978a).

2. The term's originator, Pierre Schaeffer, assigned a limited meaning to the term in his book (Schaeffer 1966). However, most use of the term today does not import the radical distinction Schaeffer made between the function of notes in traditional music and the function of sound objects in *musique concrète*.

In music theories based on the note concept, such as common-practice harmony, counterpoint, and serial music, a note's properties are assumed to be static (e.g., the pitch does not change). In traditional music, the amplitude envelope of a note was variable only through the addition of crescendo and decrescendo markings. By contrast, a sound object can have a rich and dynamic internal development, with separate envelopes on any number of parameters. The note concept also becomes problematical when one considers the spatial and spectral properties of computer-generated sound. Here again, there may be one musical event, with a definite beginning and end but of no definite pitch, that undergoes a variety of spatial and spectral mutations.

Sound Object Heterogeneity

Sound objects are *heterogeneous,* whereas notes are *homogeneous*. That is, notes are typically associated with a uniform set of properties (such as pitch, duration, amplitude, and "timbre"). Assuming, as note-based music theories do, that all notes have the same properties (i.e., every note has a specific pitch, amplitude, and so on), one can compare each note directly with another; one can even organize each property in abstraction. The consistency (such as it is) of tonal music theories is based on this uniformity. Serial and stochastic music theories that manipulate note "parameters" also depend on each note having the same properties. In contrast, objects are not so homogeneous. Two objects produced by two different sound-synthesis techniques may have an entirely different set of properties. As an example, consider two objects: *apple* and *orange*. *Apple* is a simple trumpetlike sound, with a short attack and decay envelope and a short duration (e.g., 0.5 sec). The second sound, *orange,* is a rich, long, constantly shifting, inharmonic cluster, centered at a very high frequency, which glides slowly downward, breaking up into two crackling, noisy bands of reverberated sound. Comparing the values of their common properties is like comparing a violin with a bass drum. Although the violin and the bass drum share some properties (e.g., made of wood), the only way to comprehend them is to understand the many properties they do not share.

nscor takes the principle of sound object heterogeneity as its starting point. *nscor* was worked out intuitively, but in the future it should be possible to extend the sound object concept with symbolic tools. Such tools would allow composers and researchers to manipulate sound objects more easily. Database manipulation programs that access and edit sounds according to their acoustic properties are presently in use by speech researchers (Shipman 1983).

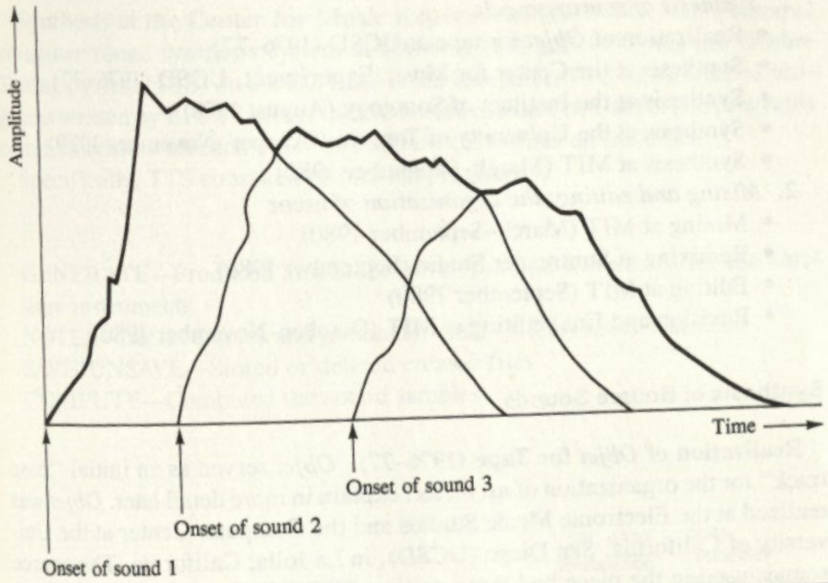


Fig. 8.1. A compound object made by mixing three sounds (derived three different ways) into one. The bold line indicates the overall amplitude envelope.

Compound Objects

Many sounds in *nscor* are created by mixing separate objects produced by different synthesis techniques into one *compound object*. This is one escape route from the predictable spectral morphologies inherent in some synthesis techniques. For example, the dynamic spectra of sounds produced by simple frequency modulation are condemned to follow the curves of the Bessel functions. In a compound sound object, a sound can begin with a harpsichord attack, then mutate into a waveshaped spectrum, and end with a reverberated granular-synthesis sound. Figure 8.1 shows a compound sound constructed from a mixture of three separate sounds.

OVERVIEW OF THE REALIZATION OF *nscor*

Having discussed the theory behind the labor, the rest of this text focuses on the practice of realizing *nscor*. The process occurred in two major stages, with several substages:

1. *Synthesis of source sounds*
   - Realization of *Objet* for tape at UCSD (1976–77)
   - Synthesis at the Center for Music Experiment, UCSD (1976–77)
   - Synthesis at the Institute of Sonology (August 1978)
   - Synthesis at the University of Toronto (October–November 1979)
   - Synthesis at MIT (March–September 1980)
2. *Mixing and editing: the organization of* nscor
   - Mixing at MIT (March–September 1980)
   - Remixing at Suntreader Studio (September 1980)
   - Editing at MIT (September 1980)
   - Revision and final editing at MIT (October–November 1980)

Synthesis of Source Sounds

Realization of *Objet* for Tape (1976–77). *Objet* served as an initial "base track" for the organization of *nscor,* as I explain in more detail later. *Objet* was realized at the Electronic Music Studios and the Computer Center at the University of California, San Diego (UCSD), in La Jolla, California. The source sounds used in the piece had three origins: a Moog III analog synthesizer, a Buchla 100 Series analog synthesizer, and computer-generated sounds from a Burroughs B6700 computer. A single interconnection "patch" generated all the analog electronic sounds in *Objet* (fig. 8.2). The patch was adapted to both the Moog and Buchla synthesizers.

Computer Sound in Objet. Portions of *Objet* were produced with a computer technique called *granular synthesis* (Roads 1978a). In this method, thousands of very short (less than 20 ms) *grains* of sound are combined to form sound spectra. The samples were computed on a dual-processor B6700 machine at the UCSD Computer Center. Digital-to-analog conversion took place at the Center for Music Experiment, UCSD.

Realization of Objet. I began by recording brief sequences of stereo sound objects, which were edited and elaborated into successively longer passages. In many studio sessions, I experimented with combinations of the passages by recording them on different tracks of a four-track tape recorder. Out of such combinations, some sections were chosen for further work. New sound objects were mixed over existing ones, or new objects were inserted, as needed.

After six months of studio work, a four-track master tape was completed. I prepared a remixing score and remixed the piece to two tracks at Beggs/American Zoetrope Recording, San Francisco. The piece was edited at the 1750 Arch Street Studio, Berkeley, in June of 1977.

Synthesis at the Center for Music Experiment (1976–77). The principal computer sound synthesis system at CME in 1976 and 1977 was the Timbre Tuning System (TTS) on a DEC PDP-11/20 computer. TTS was a suite of programs written by Bruce Leibig, then software director of CME. The programs communicated with each other through common files on the disks.

Specifically, TTS consisted of four subprograms:

GENERATE—Produced amplitude envelopes and waveforms for the software instruments
NOTES—Defined notes and parameter field (pfield) specifications
SAVE/UNSAVE—Stored or deleted created files
COMPUTE—Computed the sound samples

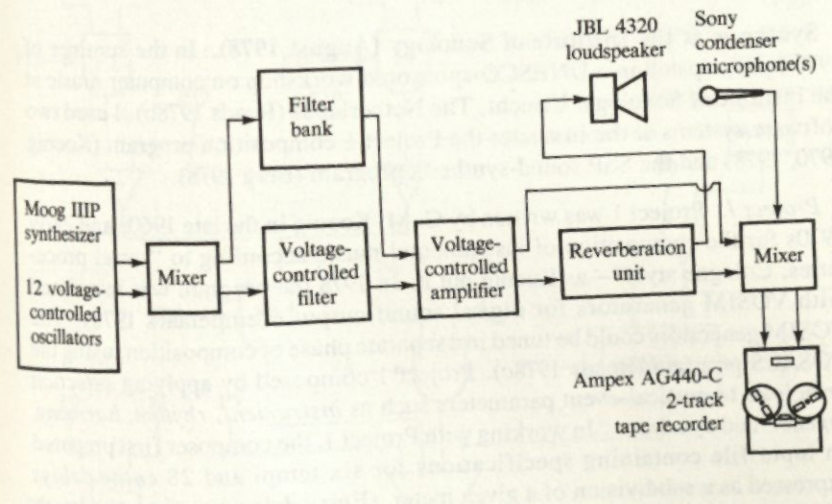


Fig. 8.2. *Additive-subtractive* patch used in the realization of *Objet*. The additive-subtractive synthesis technique involved the tuning of up to 12 audio oscillators into a single sound or frequency cluster. The balance between the individual oscillators was tuned in a mixer (this was the additive stage) and then sent to two parallel filter banks (this was the subtractive stage). The outputs from the filters were summed. The summed signal was spatially processed in three ways, which could be combined: adding reverberation, re-recording in stereo the sound played over a loudspeaker (while sometimes moving the microphones to achieve a varying sense of directionality and presence of the sound), and panning the sound (moving the sound laterally) with the mixer.

Since the PDP-11/20 was a machine of limited performance, and was not suitable for general Music V-type software synthesis,[3] TTS provided four efficient software instruments.

TTS Synthesis Tasks. The most interesting instrument in TTS was based on frequency modulation (FM) with an optional noise-modulation component (fig. 8.3). Table 8.1 is a listing of the inputs to these instruments. Despite the limitations of relying on a single instrument, quite a wide variety of sound could be produced, because I could supply both the waveform and envelope functions in addition to 28 instrument parameters. Also, some of the complexity of the sound could be embedded in the score rather than in the instrument. I usually mixed multiple, simultaneous notes to create one compound sound object. Because of memory limitations on the PDP-11/20, however, only 16 notes could be specified at a time. Thus, by far the most useful mode in working with TTS was the specification and tuning of individual sounds and short phrases. As sounds were created, they were recorded on tape and cataloged.

Synthesis at the Institute of Sonology (August 1978). In the summer of 1978 I participated in a UNESCO-sponsored workshop on computer music at the Institute of Sonology, Utrecht, The Netherlands (Roads 1978b). I used two software systems at the institute: the Project 1 composition program (Koenig 1970, 1978) and the SSP sound-synthesis program (Berg 1978).

Project 1. Project 1 was written by G. M. Koenig in the late 1960s and early 1970s for the composition of instrumental music according to "serial procedures, Cologne style"—as Koenig put it. In 1978 the program was interfaced with VOSIM generators for digital sound output (Tempelaars 1978). The VOSIM generators could be tuned in a separate phase of composition, using the VOSACS program (Roads 1978c). Project 1 composed by applying *selection principles* to musical-event parameters such as *instrument, rhythm, harmony, register* and *dynamics.*[4] In working with Project 1, the composer first prepared an input file containing specifications for six tempi and 28 *entry delays* expressed as a subdivision of a given meter. (Entry delay was concerned with whether events overlap or follow each other. In practice, the concept boils down to a numerical weighting for various sizes of chords.)

When Project 1 was run, a series of questions were posed to the user at the terminal. The full range of dialogue between the composer and the Project 1

3. The PDP-11/20 was a 16-bit computer with a total address space of 64 Kbytes of magnetic core memory and no hardware for multiplication or division; 2.4 Mwords of disk storage were available.
4. I omit a discussion of the details of Project 1 here. See Koenig (1970, 1978).

system is shown in figure 8.4. One could either waive questions by typing 0, in which case default settings prevailed, or typing 1 and responding to each question with a specific directive. I wanted a rich variety of sounds, so I created a score for 15 VOSIM instruments with vibrato (fig. 8.5). Several parameters of the VOSIM instruments could be tuned, including the center frequency on

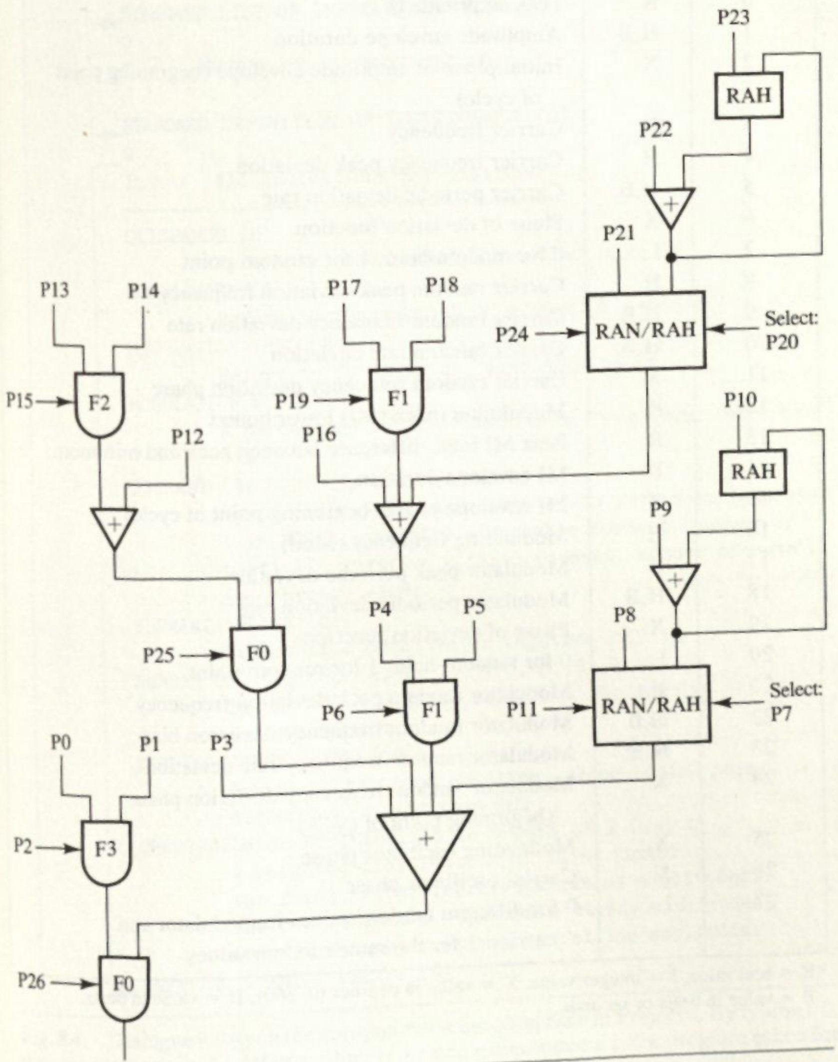


Fig. 8.3. TTS instrument 3

TABLE 8.1 Parameter Fields (pfields) for Timbre Tuning Instrument Number 3

Parameter	Type*	Explanation
0	R	Peak amplitude (F3)
1	H,B	Amplitude envelope duration
2	X	Initial phase of amplitude envelope (beginning point of cycle)
3	H	Carrier frequency
4	H	Carrier frequency peak deviation
5	H,B	Carrier periodic deviation rate
6	X	Phase of deviation function
7	I	0 for random-hold, 1 for random-point
8	H	Carrier random peak deviation frequency
9	H,B	Carrier random frequency deviation rate
10	H,B	Carrier random rate deviation
11	X	Carrier random frequency deviation phase
12	R	Modulation index (MI) lower bound
13	R	Peak MI bias, difference between peak and minimum
14	B	MI envelope duration
15	X	MI envelope phase (beginning point of cycle)
16	H	Modulating frequency (Mod)
17	H	Modulator peak periodic deviation
18	H,B	Modulator periodic deviation rate
19	X	Phase of deviation function
20	I	0 for random-hold, 1 for random-point
21	H	Modulator random peak deviation frequency
22	H,B	Modulator random frequency deviation bias
23	H,B	Modulator random frequency rate deviation
24	X	Modulator random frequency deviation phase (beginning point of cycle)
25	X	Modulating oscillator phase
26	X	Carrier oscillator phase
27	I	0 for different random values in modulator and carrier, 1 for the same random values

*R = real value, I = integer value, X = value in degrees (0–360), H = value in hertz, B = value in beats or seconds.

```
STANDARD DATA? (0)
0
1       BRANCHING BY RANDOM? (0)
        0
        1       This repeats 5 times for the parameters: instrument,
                entry delay, pitch, register, and dynamics. 0 means
                random branching is desired; 1 means branching is to
                be changed

        STANDARD LIST OF INSTRUMENTS? (0)
        0
        1       NUMBER OF INSTRUMENTS? (9)

        STANDARD DEFINITION OF INSTRUMENTS? (0)
        0
        1       All indications per instrument
                ------------------------------------
        GLISSANDO (0)
                Refers to fundamental, less than 2521 Hz
        MODULATION (0)
                Refers to fundamental, 0 = sine, 1 = random
        MODULATION FREQUENCY (0)
                Refers to fundamental, less than 64 Hz
        MODULATION WIDTH (0)
                A percentage of the fundamental frequency, less than
                51%
        FORMANT (3, 5, 7, 3, 5, 7, 3, 5, 7)
                Specify one value per instrument; refers to parameter
                t in VOSIM; positive = variable formant (number of
                partial), negative = fixed formant, number of note
                (-1 through -108)
        FORMANT CHANGE (0)
                Ranges from positive or negative 2521 Hz
        NUMBER OF IMPULSES (2, 2, 2, 5, 5, 5, 8, 8, 8)
                In the VOSIM signal, less than 16; specify one value
                per instrument
        DAMPING COEFFICIENT (50)
                Refers to previous question, slope of the pulse
                differences, less than 101
        CRESC/DECRESC DISTRIBUTION IN GROUPS OF 3 (0 0 1)
                A weighted distribution of three numbers:
                the first number represents rising amplitudes;
                the second number represents steady amplitudes;
                the third number represents falling amplitudes
        DURATION = ENTRY DELAY? (0)
```

Fig. 8.4. Dialogue between the computer and the composer in Project 1. By typing a 0, the composer selected a default value. If the composer typed a 1, the program asked for more specifications.

```
       0
       1       DURATIONS PER INSTRUMENT (0)
               ----------------------------
               -2 staccato
               -1 duration less or equal to entry delay
                0 duration equals entry delay
                1 durations greater or equal to entry delay
                2 duration independent

   STANDARD REGISTERS
   0
   1       NUMBER OF REGISTERS (4)

           REGISTER NUMBERS (3, 4, 5, 6)
           ----------------------------
           Octave 1 =     16 Hz
           Octave 2 =     33 Hz
           Octave 3 =     65 Hz
           Octave 4 =    131 Hz
           Octave 5 =    262 Hz
           Octave 6 =    523 Hz
           Octave 7 =   1047 Hz
           Octave 8 =   2093 Hz
           Octave 9 =   4186 Hz

   STANDARD DYNAMICS (0)
   0
   1       8 DYNAMIC VALUES: (1, 2, 3, 4, 5, 6, 7, 8)
           ------------------------------------------
           1 is very soft, 8 is very loud

   PRINT STRUCTURE NUMBER
   -1
    0, N   Print structure N
   PLAY LINES ... THROUGH ...
   -1
    0, M, N        Play lines M through N
   PRINT DURATION TABLE OF STRUCTURE NUMBER
   -1
    0, N   Print duration table N
   RECALCULATE IN STRUCTURE NUMBER
   -1
    0, N

END OF PROGRAM (0)
0        Return to monitor level
1        Restart Project One
```

Fig. 8.4. *Continued*

```
VERSION  9999
REPETITION RANGE  1
NUMBER OF CHORDS

STRUCTURE  1

COMBINATION

PARAMETER   PROGRAM

INSTRUMENT   6
RHYTHM       3
PITCH        3
REGISTER     1
DYNAMICS     5

     INSTR    RHYTHM      HARMONY            SEQUENCE   REGISTER     DYNAMICS

           TEMPO  ED FERMATE

  1  *13  90.0 * 1/5     * F  F# A#          3 1 2      2 7 4        * FFF
  2   13         1/4       G# A  C#          3 2 1      9 8 3          FFF
  3   13         2/5       B  C              1 2        6 5            FFF
  4   13         3/8       E  D              2 1        9 2            FFF
  5   13         3/5     * A# B  D# G  G#    2 1 5 3 4  3 7 4 3 9      FFF
  6  * 7         2/3       C  C# D           3 1 2      2 6 1        * MP
  7    7       * 1/1       F# E  F           3 1 2      8 3 7          MP
  8    7         1/7     * C  C# F           2 1 3      3 1 8          MP
  9    7         3/8       D# E              1 2        7 1            MP
 10  *11         4/5       G#                           4            * F
 11   11         1/4       F#                           5              F
 12   11         1/5       G                            2              F
 13   11       * 3/8       B  A  A#          2 3 1      3 7 5          F
 14   11         1/9       D                            3            * FF
 15  * 2         1/4     * A  A# D           3 1 2      8 2 6          FF
 16    2         1/7       F# G  B           3 1 2      1 3 9          FF
 17    2         0/0 N     D#                           6              FF
 18    2         2/3   1   E  G# C  C#       3 2 4 1    7 2 3 8      * PPP
 19    2  70.0 * 1/8     * E  F              1 2        3 6            PPP
 20    2         2/5       G# C# C  A        4 3 2 1    9 1 4 3        PPP
 21  *10         0/0 N     D#                           7              PPP
 22   10         1/8       D  B              2 1        2 8            PPP
 23   10         3/8       F# G  A#          1 2 3      2 6 7        * P
 24   10         0/0 N 1 * F                            4              P
 25   10  30.0 * 2/3       F# A  E  D#       1 4 3 2    5 6 3 1        P
 26   10         0/0 V     C                            5              P
 27  * 3         1/8       G                            4            * PP
 28    3         0/0 N     G#                           2              PP
 29    3         1/4       B                            3              PP
 30    3         5/8     * F# G  A# G# A     4 3 5 1 2  1 6 9 4 6      PP
 31    3       * 5/8       C  F  E           3 2 1      3 2 6        * MF
```

Fig. 8.5. Beginning of the second score generated by Project 1

which the VOSIM formant focused, specified as a harmonic of the fundamental frequency. To avoid simple harmonic sounds, I stipulated the relatively distant ninth harmonic as the formant for this study. Specifying 15 impulses in the VOSIM signal (with no damping) ensured a rich spectrum.

SSP. G. M. Koenig's Sound Synthesis Program (SSP) was a means of building up waveforms from time and amplitude point specifications (Berg 1978). After specifying an initial data set, SSP provided a number of *selection principles* that allowed me to elaborate this set into sound objects and phrases. Since the selection principles were based on serial, logical, and random operations to generate sound (rather than traditional signal-processing operations such as oscillators and filters), SSP was well suited to the production of jagged waveforms and rich, amplitude-modulated textures, juxtaposed in series.

The SSP program ran interactively on the DEC PDP-15 computer. Because of the direct and uncomplicated method of sound synthesis, the typical sample computation time was a few seconds.

Synthesis at the University of Toronto (October–November 1979). In the autumn of 1979 I was a visiting scholar at the University of Toronto, affiliated with the Structured Sound Synthesis Project (SSSP) led by William Buxton.

The SSSP Digital Synthesizer. The SSSP digital synthesizer (Buxton et al. 1978) was a 16-voice system with four synthesis methods in hardware:

- Fixed-waveform additive synthesis
- Frequency modulation (FM)
- Nonlinear distortion or waveshaping
- VOSIM

Eight different waveforms could be stored in the wavetable memory of the synthesizer. These could be used as amplitude envelopes or audio waveforms in oscillators.

Software. The SSSP software was a powerful complement to the synthesizer. The majority of this software was graphics-based, allowing a musician to do a great deal of work interactively, without the heavy burden of typing endemic to other computer music systems (Buxton et al. 1979). Typing was avoided through reliance on a Bit-Pad digitizer and hand-held pointing device, in conjunction with window-and-menu-oriented graphics software (fig. 8.6). At any point in the interaction, the user was presented with a set of optional actions, displayed as menus in various windows of the screen. By pointing at a

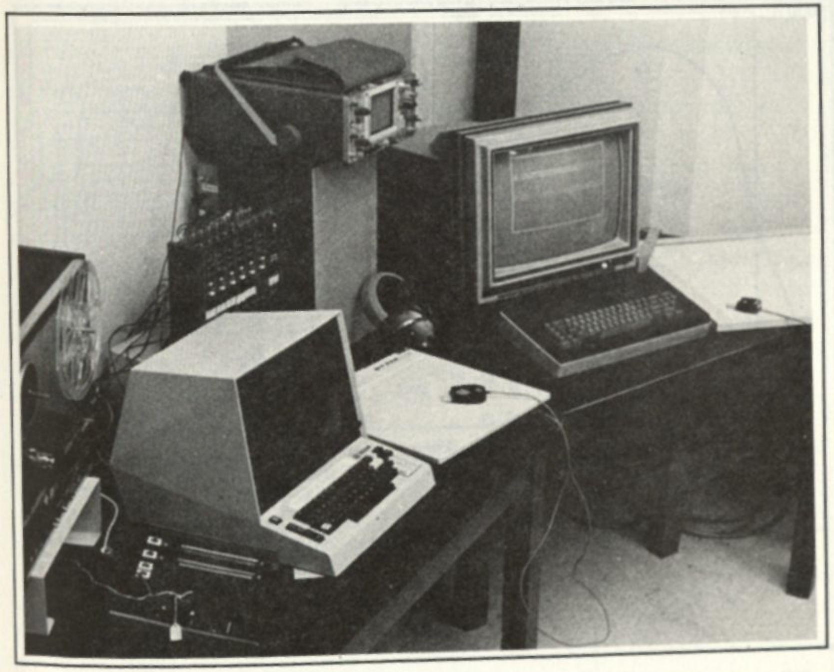


Fig. 8.6. The SSSP Laboratory workstation in 1978. Interaction tools included (from left): a slider mixer and switch box, a low-resolution terminal with a digitizer tablet and pointing device, a high-resolution vector display terminal with a digitizer tablet and pointing device. An oscilloscope, loudspeakers, an amplifier, and tape recorders were also provided.

selected item on a menu and pressing a button on the pointing device, the user selected the menu item and the operation associated with it was performed.

I made extensive use of several major software packages in generating sounds in *nscor*, including:

- SCRIVA—a graphics-based score editor (fig. 8.7)
- OBJED—a graphics-based sound object editor that allowed the composer to create and edit waveforms, tune digital sound objects, and test the sounds in a variety of musical contexts (fig. 8.8)
- ORCH—a graphics-oriented program that allowed the composer to select various musical events displayed on the screen and attach an instrument to them
- PLAY—a sound playback system with many options

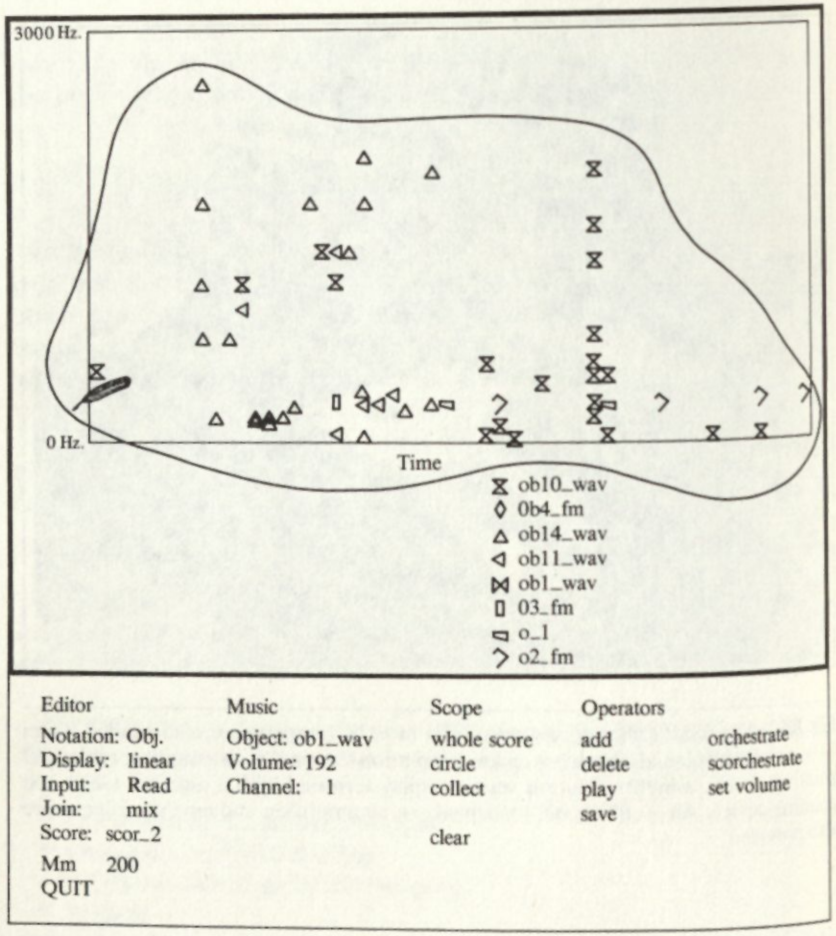


Fig. 8.7. Screen image of a score fragment produced by SCRIVA in "symbolic object" format (no staves). All of the sound objects have been circled. This encirclement defines the scope of the next operation to be performed—for example, "Play."

Besides these main programs, a collection of one-line commands existed for transformation of subscores or scores. Some commands used in my work included:

- MIX—mix scores together to produce a new score
- RETRO—create a new score as the retrograde of an existing score

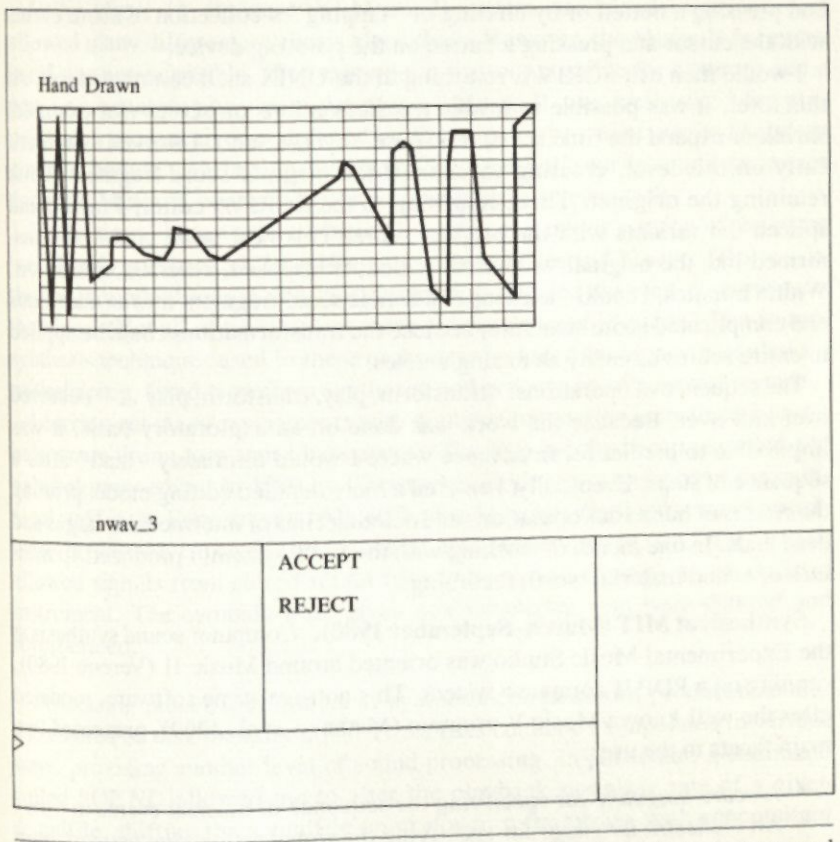


Fig. 8.8. Hand-drawn waveshaping function "nwav_3" made with the OBJED sound object editor

- SETVOL—set the volume of a score
- SPLICE—splice scores in sequence
- TRANSP—transpose the pitches to another frequency base
- TSCALE—shrink or expand the time scale of a score
- SETCHAN—set the number of channels of sound output

Synthesis Methodology. At first, I created small scores with the SCRIVA score editor. (The size of the scores was limited by the size of the frame buffer of the graphics hardware. Scores were stored in the frame buffer as lists of vectors used to reconstruct the graphic image.) Second, I tuned various objects using OBJED and orchestrated the score with the tuned instruments. This was all accomplished with graphics techniques, by pointing at specific score events and pressing a button or by circling or "tapping" a collection of score events with the cursor and pressing a button on the pointing device.

I would then exit SCRIVA, returning to the UNIX shell command level. At this level, it was possible to invoke transformations on scores (for example, shrink or expand the time scale, transpose, reverse, etc.). I worked systematically on this level, creating variants of the original score fragment while retaining the original. Then, returning to the SCRIVA editor, I mixed and spliced the variants with the original score. This new score could be transformed like the original was—time-scaled, transposed, reversed, and so on. Within minutes, I could "telescope" a simple score fragment into an enormous and complicated score structure, because the transformations could be applied to entire scores as easily as to single notes.

The sequence of operations: "transform, play, transform, play . . ." occurred over and over. Because the work was done on an exploratory basis, it was impossible to predict far in advance where I would ultimately "land" after a sequence of steps. Eventually, I entered a more detailed editing mode, pruning the results of numerous operations and deleting files of intermediate stages and dead ends. In one month of working with the SSSP system, I produced 40 minutes of sound material worth recording.

Synthesis at MIT (March–September 1980). Computer sound synthesis at the Experimental Music Studio was oriented around Music 11 (Vercoe 1980), running on a PDP-11 computer system. This non-real-time software, modeled after the well-known Music V program (Mathews et al. 1969), presented two main facets to the user:

- A *score language* for specifying the sequence of musical events in time
- An *orchestra language* for specifying the signal processing network used to synthesize sound

The score was simply a list of instrument names, starting times, durations, and parameter fields (called *pfields*) that supplied numerical values to the orchestra. An orchestra was designed out of individually defined *instruments*. The instruments were signal processing networks constructed by linking *unit generators* with audio and control signals.

Unit generators were the primitive components of the Music 11 orchestra language. They included the following types:

Oscillators	Envelope generators
Soundfile inputs	Random-signal generators
First-order filters	Delay units
Second-order recursive filters	Reverberators
Impulse generators	Spatial-output generators

Using Music 11. Because the Music 11 system was based on software, it allowed many different synthesis algorithms. However, the Music 11 language itself was not extensible. (For example, it was not possible for a user to add a new unit generator or a new data type.) Music 11 was well suited for pinpointing a preconceived sound and synthesizing it, provided one had an acute grasp of how to translate a sound heard in the mind's ear into unit-generator code and Music 11 control structures—not always an easy task.

By the time I began to use Music 11 for *nscor,* a major portion of the piece was already constructed. I used 8 small orchestra files and 12 score files to synthesize individual sounds and short fragments, tailored to specific moments in the piece. (Not all of these scores were ultimately used in *nscor.*) The sound-synthesis techniques used in these orchestras included frequency modulation, waveshaping, fixed-waveform synthesis, and processing of concrète sound.

I carried out many experiments with digitized percussion instruments (cymbals, snare drum, tom-toms, bass drum). These sounds had been recorded and painstakingly edited in 1980 by Christopher Fry, then a member of the technical staff at the Experimental Music Studio. In *nscor,* digitized cymbal sounds were processed through the Music 11 unit generator SOUNDIN, which allowed signals from stored sound files to be passed through a user-defined instrument. The cymbals were given new envelopes, and were filtered and reverberated.

Processing Soundfiles. Music 11 synthesis computations produced soundfiles stored on disk memory units. These files could be manipulated in various ways, providing another level of sound processing. In particular, a command called SOUND allowed me to alter the playback sampling rate of a given soundfile, shifting the soundfile up or down in frequency, with concomitant stretching and shrinking in time.

Mixing and Editing: The Organization of *nscor*

Mixing at MIT (March–September 1980). By March 1980, many of the basic sound materials to be used in fabricating *nscor* had been generated and cataloged (see Appendix B). The task before me then was to organize these sound objects into a functioning composition. Upcoming concerts in the autumn placed a six-month deadline on the compositional labor. As a practical consideration, I decided to work with the new digitally generated sounds in the context of the composition *Objet*. Using a four-channel tape recorder, I would record *Objet* on two tracks, leaving two tracks for new material. I planned to rearrange and delete major portions of *Objet* as new material was integrated into the piece.

Phrase Structure in nscor. *Objet* is a continuum, without a rest. In *nscor*, I hoped to achieve a more exaggerated phrase structure than in *Objet*. The obvious device for delineating phrase structure was the insertion of silences between phrases. In *nscor*, phrases have an inner structure, with an opening, a development, and a closure. The notion of "development" in *nscor* usually consists of a contrapuntal sequence involving objects of similar synthesis origin (e.g., waveshaping).

The Realization. The first realization step, in March 1980, was to record *Objet* on two channels of a four-channel tape machine. The next step was to alter the original syntax of *Objet* by separating out self-contained sections and finding points where new material could be inserted. I began to layer new digital sounds onto the four-channel tape. Up to two sound streams could be layered in parallel with *Objet* before it was necessary to transfer the four signals to the two-channel tape recorder. Such sounds could be mixed back onto two channels of the four-channel machine for more complicated textures. Also at this stage I began work on a new beginning section for *nscor* built out of digital sound objects.

Partitioning the Tape. After initial editing steps, I partitioned the piece into 17 sections separated by leader tape. The point of partitioning was to concentrate further editing work within a section. It still required many reordering experiments to sort out and reorder a sequence of incorrectly ordered objects in a section.

Editing the Tape. Much of the work from late March to August of 1980 was a process of editing: inserting, deleting, and modifying. Each time a new object was added to the piece, the functions of the objects near it were affected. As new objects were mixed in or inserted, more and more old objects from *Objet* seemed out of place and were deleted. As planned, there is little left of *Objet* in *nscor*.

Selection and Adjustment. In the final stages of organization, the process of composition became one of honing rough edges, balancing levels with the mixer, and snipping loose ends—selecting only those events whose properties contributed to the morphology of a particular passage. When an entire phrase had been synthesized whole (many of the sounds from the Toronto studio were organized this way), it was occasionally necessary to first delete intermediate events to compress the phrase or add foreign events to expand it. The four-channel master tape of *nscor* was completed in early September 1980, in preparation for an immediate remix process.

Remixing *nscor* at the Suntreader Studio (September 1980). I planned the production of the four-channel master tape as an intermediate phase in the realization of *nscor*. To obtain the highest possible sound quality, I wanted to remix the four-channel tape to a two-channel master at a professional recording studio. The goals of the remixing included:

1. High-quality, low-noise mixing of the four-channel version to a standard two-channel performance format
2. Fine adjustment of dynamics
3. Spatial distribution of the sounds in two channels
4. Addition of reverberation and short signal delays at selected points
5. Adjustment of timbre through fine-tuned equalization of the spectrum at selected points

Items 2 through 5 were planned to add "finish" to a raw source tape. The adjustments possible in the mixing studio were to add nuance and color not achievable by any other means. In September 1980 I visited the Suntreader Studio in Vermont to remix *nscor*. Before the remixing session, I prepared written notes that included specific directives for each section (fig. 8.9). These notes served as a starting point for creative mixing. Additional adjustments made in the studio are not reflected in the notes.

The mixing was carried out systematically. Each section was auditioned; then the mixer was set up for that section. The mix was rehearsed, and various takes of the mix of a section were recorded. After a day of work, I had a two-track tape with 17 separate sections of *nscor* recorded on it.

Editing at MIT (September 1980). The primary goal of the editing was to delicately reconnect the 17 remixed sections, separated by silences and outtakes, into a finished composition. The work was not as straightforward as one might assume. First, I had to eliminate minor glitches (clicks) and other audio flaws that remained as the residue of hundreds of hours of studio work. A variety of splices were created to cope with them, and the piece was successfully deglitched.

In addition, because each of the 17 sections was mixed separately, the audio level of the end of an antecedent section did not always match the beginning of the consequent section. I adopted two surgical solutions to this problem. First, instead of connecting the two endpoints fixed on the master tape, I backtracked slightly on each tape section to a point of better level balance. I then prepared a long diagonal splice, which smoothed the transition between the two new endpoints. The second solution involved additional recording, when backtracking was deemed too radical. The section fragments were re-recorded on tape at a

Section 6 [:50]

Material: Starts with processed cymbal object
Amplitude: Slightly lower than the 4-track tape
Balance: Lower channels 1 and 2 slightly
Equalization: Flat
Reverberation: Medium wet
Delay: None

Section 7 [:58]

Material: Starts with waveshaped sound mass
Amplitude: Start at very low level, gradually increase
Balance: Lower channels 1 and 2 slightly
Equalization Add 3 dB at 12.5 kHz
Reverberation: Dry
Delay: None

Section 8 [:48]

Material: Waveshaped phrases
Amplitude: Expand the dynamic range of the envelopes on the tape
Balance: Equal
Equalization: Flat
Reverberation: Selectively add reverberation, especially to the final crescendo
Delay: None

Section 9 [:39]

Material: Analog synthesizer textures, then granular synthesis
Amplitude: Increase amplitude slightly at the crescendo
Balance: Equal
Equalization: On channels 1 and 2, subtract 4 dB at 7.5 kHz
Reverberation: Dry
Delay: None

Section 10 [:23]

Material: Additive synthesis textures
Amplitude: Start at moderately high amplitude; as tone in channels 1 and 2 enters, bring to a crescendo
Balance: Lower channel 3 slightly
Equalization: Flat
Reverberation: Medium wet
Delay: None

Section 11 [:53]

Material: Granular-synthesis textures
Amplitude: Keep low
Balance: Lower channels 3 and 4 at the beginning and gradually raise them
Equalization: Flat
Reverberation: Dry
Delay: Gradually add 50 ms delay to high-frequency grains, then fade the delay out

Fig. 8.9. Excerpt from the remix notes for *nscor,* sections 6 to 11

more balanced level and then spliced. (The sound quality of the master was very good, so re-recording did not pose a major noise-buildup problem.) This produced a smooth and seamless transition to the new section.

Juxtaposition plays an important structural role in *nscor,* and not all of the section transitions were meant to be seamless. Using these same techniques, I articulated some section transitions as sharply as possible. The work was completed on 4 September 1980. The resulting spliced version of *nscor* reduced the length of the piece from 11 min, 9 sec (combined length of the 17 sections on the four-channel master) to 10 min, 41 sec. To meet a deadline, the September version of *nscor* was shipped to Varese, Italy, for performance at the second "Concorso Internazionale 'L. Russolo.'"

Revision and Final Editing at MIT (October–November 1980). The revision surgery began with the same techniques of splicing and re-recording as before, applied toward structural problems remaining in the piece. For example, I wanted to add considerable new material to the work while cutting some passages wholesale; this involved reworking certain sections to make them all fit together properly. Problematical passages included the opening section, the ending, certain long stretches of additive synthesis material in section 12, and remaining fragments of *Objet* that had grown out of place as the rest of *nscor* developed around them.

Many resources were brought to bear on these problems. New sounds were synthesized using the Music 11 software, and the four-channel tape recorder was again used for the manipulation of compound sounds. Classical tape looping and splicing techniques were extended to their limit to produce the massed sounds of the ending.

The resulting November version of *nscor* lasts 8 min, 45 sec, in contrast to the September version of 10 min, 41 sec. Although much new material was added to the piece, the overall length was cut by just under two minutes. The American premiere of *nscor* took place at the 1980 International Computer Music Conference, Queen's College, New York, 13 November 1980.

CONCLUSION

To compose with sound objects means to select one object from a vast palette, to shape its evolution in time, and to combine it with other objects. The tools we use to select, shape, and combine sound objects unavoidably set the boundaries of the music we can create.

This text reveals the simultaneously "advanced" and yet also limited state of computer music tools. They were advanced, because they offered better sound quality and a wider selection of sounds than previous systems. But they were also limited—and even primitive in some ways. To venture beyond this state of affairs, a new generation of more flexible systems is needed. The integration of real-time sound synthesis and sound mixing is essential to the production of complex sound objects. One shouldn't have to work in a separate mixing studio.

As it becomes possible to store large sound catalogs, the need to generate every sound anew should disappear. Part of the burden of the musician-machine interface can shift from the invention and reinvention of synthesis instruments to techniques for symbolically accessing and manipulating cataloged sound objects—creation of high-level structure and process.

The tremendous range of material explored in *nscor*—from metallic, inharmonic splashes to smooth and distant textures, cloudbursts of sound and shimmering sound rays—forces new ways of thinking about composition. Although current trends might take decades to fully unfold, we stand in the midst of an ongoing expansion of musical thought to encompass the sounds, structures, and processes made possible in this century.

ACKNOWLEDGMENTS

I am indebted to the following persons for sponsoring my work and for developing many of the tools I used to create *nscor:* Bruce Leibig at UCSD; William Buxton at the University of Toronto; G. M. Koenig at the Institute of Sonology; and Barry Vercoe, Roger Hale, and Christopher Fry at the MIT Experimental Music Studio. In addition, I greatly appreciate the comments of John Strawn and Stephan Kaske on a draft of this manuscript.

REFERENCES

Berg, Paul. 1978. *A user's manual for SSP.* Utrecht: Institute of Sonology.

Buxton, W., E. A. Fogels, G. Fedorkow, L. Sasaki, and K. C. Smith. 1978. An introduction to the SSSP digital synthesizer. *Computer Music Journal* 2(4):28–38. Reprinted in Curtis Roads and John Strawn, eds. *Foundations of computer music.* 1985. Cambridge: MIT Press.

Buxton, W., R. Sniderman, W. Reeves, S. Patel, and R. Baecker. 1979. The evolution of the SSSP score editing tools. *Computer Music Journal* 3(4):14–25, 60. Reprinted in Curtis Roads and John Strawn, eds. *Foundations of computer music.* 1985. Cambridge: MIT Press.

Grey, John M. 1975. An exploration of musical timbre. Ph.D. diss., Department of Psychology, Stanford University. Department of Music Report STAN–M–2.

Koenig, G. M. 1970. Project 1. *Electronic Music Reports* 2. Utrecht: Institute of Sonology. Reprinted 1977 by Swets and Zeitlinger, Amsterdam.

Koenig, G. M. 1978. Description of the Project 1 programme. Utrecht: Institute of Sonology.

Mathews, Max V., with Joan E. Miller, F. Richard Moore, John R. Pierce, and Jean-Claude Risset. 1969. *The technology of computer music.* Cambridge: MIT Press.

Risset, Jean-Claude. 1968. An introductory catalogue of computer synthesized sounds. Murray Hill, N. J.: Bell Telephone Laboratories.

Roads, Curtis. 1978a. Automated granular synthesis of sound. *Computer Music Journal* 2(3):61–62. Revised and updated version published as "Granular synthesis of sound" in Curtis Roads and John Strawn, eds. *Foundations of computer music.* 1985. Cambridge: MIT Press.

Roads, Curtis. 1978b. A report on the UNESCO workshop on computer music at Aarhus, Denmark. *Computer Music Journal* 2(3):30–32. Reprinted in M. Battier and B. Truax, eds. *Computer music/compositional musicale par ordinateur*, xi–xxix. 1980. Ottawa: Canadian Commission for UNESCO.

Roads, Curtis. 1978c. An interview with G. M. Koenig. *Computer Music Journal* 2(3):11–15, 29. Reprinted in Curtis Roads and John Strawn, eds. *Foundations of computer music.* 1985. Cambridge: MIT Press.

Schaeffer, P. 1966. *Traité des objets musicaux.* Rev. ed. 1977. Paris: Editions du Seuil.

Shipman, D. 1983. SpireX: Statistical analysis in the SPIRE acoustic phonetic workstation. Presented at the IEEE Conference on Acoustics, Speech, and Signal Processing, Boston.

Tempelaars, S. 1978. The VOSIM oscillator. Utrecht: Institute of Sonology.

Vercoe, B. 1980. Music 11 reference manual. Cambridge: MIT Experimental Music Studio.

APPENDIX A: STUDIO EQUIPMENT USED IN THE REALIZATION OF *nscor*

Electronic Music Studios (2), UCSD, La Jolla: 1976

Moog IIIp synthesizer with Bode ring modulator
2 UREI 565 filter sets
Ampex AG440-C 4-track tape recorder
2 Ampex AG440-C 2-track tape recorders
2 Sony electret microphones
2 Macintosh amplifiers
4 JBL 4320 loudspeakers

Buchla 100 series synthesizer with 4 sequencers
1 Sony 2-track tape recorder
2 Teac 2-track tape recorders
Macintosh amplifier
2 JBL 4311 loudspeakers

Center for Music Experiment, USCD, La Jolla: 1976

DEC PDP-11/20 computer with 64 KB of memory
2 DEC RK05 disk drives (1.2 MB each)

2 16-bit DACs
14-bit ADC
Tascam mixing console
Ampex AG440-C 4-track tape recorder
Ampex AG440-C 2-track tape recorder
Sony 4-track tape recorder
4 JBL 4311 loudspeakers
2 amplifiers

Beggs/American Zoetrope Recording, San Francisco: 1977

Auditronics mixing console
Ampex MM1100 16-track tape recorder
Ampex AG440-C 4-track tape recorder
Ampex AG440-C 2-track tape recorder
3 JBL 4311 loudspeakers
2 amplifiers

1750 Arch Street Studio, Berkeley: 1977

MCI mixing console
3M M79 8-track tape recorder
Ampex AG440-C 2-track tape recorder
Revox HS77 2-track tape recorder
4 JBL 4310 loudspeakers
2 amplifiers

Institute of Sonology, Utrecht: 1978

2 DEC PDP-15 computers
2 DEC disk drives
2 DEC DECtape units
Teletype terminal
DEC VT05 terminal
2 12-bit DAC units
6 VOSIM generators
Plotter
Revox A700 tape recorder
4 loudspeakers
2 amplifiers

SSSP Laboratory, Toronto: 1979

DEC PDP-11/45 computer with 256 KB of memory
2 DEC disk drives (60 MB each)
Three Rivers Graphics Processor

Hewlett-Packard vector display terminal
Bit-Pad digitizer tablet
SSSP digital synthesizer
Teac mixer
Teac 4-track tape recorder
Teac 2-track tape recorder
Advent loudspeakers
Advent amplifier

MIT Experimental Music Studio, Cambridge: 1980

DEC PDP-11/55 computer with 192 KB of memory
2 DEC RL02 disk drives (10 MB each)
DEC PDP-11/34 computer with 64 KB of memory
Control Data Corporation 300-MB disk drive
Analogic AP400 array processor
Imlac vector display terminal
6 DEC VT52 terminals
Scully 280B 4-track tape recorder
Otari 4-track tape recorder
Otari 2-track tape recorder
4 15-bit DACs
Tascam mixer
4 Klipschorn loudspeakers
2 Crown DC300A amplifiers

Suntreader Studio, Vermont: 1980

Automated Processes mixing console
Studer A-80 24-track tape recorder
3M M79 4-track tape recorder
Studer A-80 2-track tape recorder
Eventide digital delay unit
UREI 565 filter set
Acoustic echo chamber
EMT 240 gold foil stereo reverberation unit
4 Crown DC300A amplifiers
Altec 9846 loudspeakers (biamplified) with UREI crossovers

APPENDIX B: SOURCE TAPES USED IN THE REALIZATION OF *nscor*

Granular synthesis examples (1975). One-minute granular-synthesis sections. AGS program and Music V. Burroughs B6700 computer system. UCSD.

TTS-1 (1976). Noise-modulated FM textures. Additive-synthesis textures. Timbre Tuning System software. PDP-11 computer. Center for Music Experiment, UCSD.

TTS-2 (1976). FM noise bands. Additive-synthesis tones. Timbre Tuning System software. PDP-11 computer. Center for Music Experiment, UCSD.

TTS-3 (1976). FM noise bands. Timbre Tuning System software. PDP-11 computer. Center for Music Experiment, UCSD.

TTS-4 (1976). High-frequency FM glissandi. FM textures with varying frequency curves. Additive-synthesis tones. Timbre Tuning System software. PDP-11 computer. Center for Music Experiment, UCSD.

Source 77 (1977). Stereo granular-synthesis sections. Digital FM gongs. Analog FM sounds. Timbre Tuning System software. PDP-11 computer. Moog IIIp synthesizer. Center for Music Experiment, UCSD, and Electronic Music Studio, UCSD.

Objet (1977). Master tape of the composition, produced at Beggs/American Zoetrope Recording.

Utrecht sources (1978). Project 1 experiments with output from 6 VOSIM generators. SSP textures from PDP-15 computer. Institute of Sonology.

Toronto Source 1 (1979). Short and fast score sections using waveshaping. Slowed version of score file *scor_101* using fixed-waveform synthesis. SSSP digital synthesizer. PDP-11 computer. University of Toronto.

Toronto Source 2 (1979). Experiments with the score file *tscor*, using the waveshaping technique with hand-drawn waveshaping functions. SSSP digital synthesizer. PDP-11 computer. University of Toronto.

Toronto Source 3 (1979). Experiments with score file *scor_y4*, using waveshaping. SSSP digital synthesizer. PDP-11 computer. University of Toronto.

Toronto Source 4 (1979). Fixed-waveform textures with waveforms built by sine summation. Score file *nscor_101* at tempo 1000. *nscor_101* transposed to 230 Hz and played at tempo 1000. *scor_y1* played with fixed-waveform instruments at tempo 200. *scor_y1* and *scor_y4* played at tempo 3000. SSSP digital synthesizer. PDP-11 computer. University of Toronto.

Music 11 Sources (1980). Additive-synthesis textures. Reverberated additive-synthesis textures. Reverberated FM noise bands. Delayed noise-burst textures. Processed percussion sounds. Music 11 software. PDP-11 computer. MIT Experimental Music Studio.

IANNIS XENAKIS

9

MUSIC COMPOSITION TREKS

Iannis Xenakis

MUSICAL UNIVERSES

The universes of music—classical, contemporary, pop, folk, traditional, avant-garde, etc.—seem to form units in themselves, sometimes closed, sometimes interpenetrating. They present amazing diversities, rich in new creations but also fossilizations, ruins, wastes, all in continuous formation and transformation like the clouds, so differentiated and ephemeral.

This is explained by the proposition that music is a sociocultural phenomenon subordinated to a given instant in history. However, one can distinguish parts that are more invariant than others and thus form materials of hardness and consistency, resulting from diverse epochs of civilizations, materials that move in space, created, hurled, and driven by currents of ideas, clashing, influencing, annihilating, and fecundating each other.

But of what essence are these materials made? This essence is the intelligence of man solidified in a way: intelligence that seeks, questions, infers, reveals, and foresees at all levels. Music and the arts in general seem necessarily to be a solidification, a materialization of this intelligence. Naturally, this intelligence, though humanly universal, is diversified by the individual, by talent that distances one individual from the others.

Talent is therefore a kind of qualification, a gradation of the vigor and richness of intelligence. For intelligence is, fundamentally, the result, the expression of billions of exchanges, of reactions, of transformations of energy in the cells of the brain and the body. One could, taking astrophysics as an image, say that intelligence is the form taken by the minimal acts of cells in their condensations and their movement, as happens with the particles of the suns, planets, galaxies, and clusters of galaxies born of or turned back into cold interstellar dust. This image, however, is reversed (at least on one level) because this cold dust, in condensing, becomes hot, contrary to the intelligence that is a cold result of the exchanges between the hot cells of the brain and of the body—a "cold fire."

It results that music is a strong condenser, perhaps stronger than the other arts. It is for this reason that I give a table of comparisons between certain conquests of music and several realizations of mathematics taught to us by history (see Appendix A). This table shows one of the paths that music has taken since

its origin (since antiquity) and that it has held with remarkable fidelity through millenia with a strong acceleration in the twentieth century. This proves that, far from being a fashion, this faculty of condensation toward abstraction is of a profound nature that belongs to music no doubt more than to other arts. Consequently, it seems that a new kind of musician is necessary, that of the *artist-conceiver* of free and abstract new forms, tending toward complications and generalizations on several levels of sound organization. For example, a form, a construction, an organization built on Markovian chains or on a complex of interlocked probability functions may be carried over simultaneously on several levels of musical micro-, meso-, and macrocompositions. One can also extend this remark to the visual domain—for example, in a *spectacle* made out of laser beams and electronic flashes like those of the *Polytope of Cluny* and the *DIATOPE* of the Centre Georges Pompidou (fig. 9.1).

Nothing prevents us from foreseeing from now on a new relationship between the arts and sciences, particularly between the arts and mathematics, in which the arts would consciously "pose" problems for which mathematics ought to and must forge new theories.

The artist-conceiver will have to possess knowledge and resourcefulness in domains as varied as mathematics, logic, physics, chemistry, biology, genetics, paleontology (for the evolution of forms), human sciences, and history—in

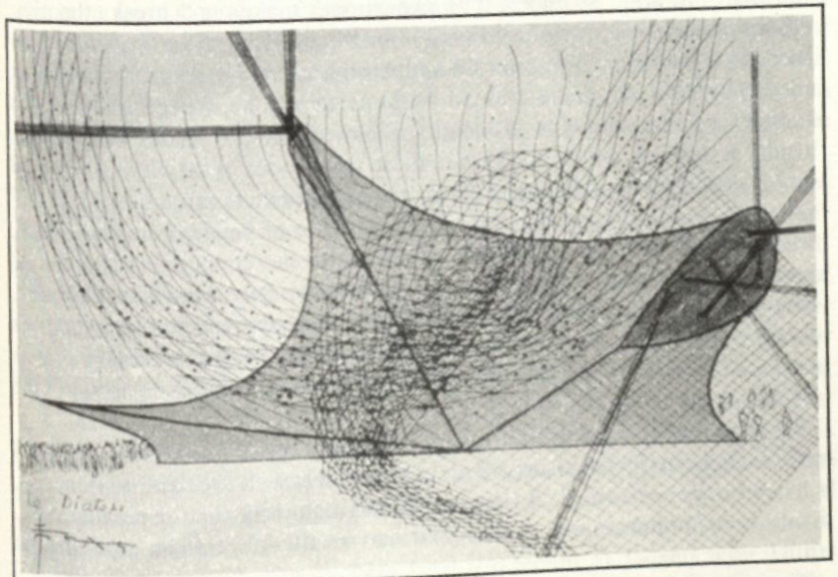


Fig. 9.1. Author's handmade lithograph of the *DIATOPE*

short a kind of universality, but a knowledge founded, guided, oriented by and toward forms and architectures. It is also time to found a new science of *general morphology* that will deal with forms and architectures of these diverse disciplines, studying their invariant aspects and the laws of their transformations, which have sometimes lasted millions of years. The backdrop of this new science should be the real condensations of the intelligence—that is, the abstract approach cleared of the anecdotes of our senses and habits. For example, the formal evolution of the vertebrae of dinosaurs is one of the paleontological documents to add to the dossier of the science of forms.

Let us plunge now into the fundamental system on which art rests. Art has something in the nature of an inferential mechanism, which constitutes the ground on which move all the theories of mathematical sciences, physics, and those of living beings. Indeed, the games of proportions reducible to games of numbers and metrics in architecture, literature, music, painting, theater, dance and so on—games of continuity, of proximity, in- or outside of time, of topologic essence—are all made on the terrain of the inference, in the strict logical sense. Besides this terrain exists the experimental mode that challenges or confirms the theories created by the sciences, including mathematics. Since the development of non-Euclidean geometries and the theorem of Gödel, mathematics has also proven to be an experimental science, but over a longer time span than the other sciences. The experiment makes and breaks theories, without pity and without consideration for them. Now, the arts are also governed, in a manner still more rich and complex, by the experimental mode. Indeed there are not, and without a doubt never will be, objective criteria for absolute and eternal truths of validity of a work of art, just as no scientific "truth" is definite. But in addition to these two modes of activity—inferential and experimental—art lives in a third, that of immediate *revelation* that is neither inferential nor experimental. The revelation of beauty is made at once, directly, to the person ignorant of art just as to the connoisseur. Revelation makes the force of art and, it seems, its superiority over the sciences because, living in the two dimensions of the inferential and experimental, art possesses this third possibility, the most mysterious of all, the one that makes the objects of art escape any aesthetic science all the while indulging in the caresses of the inferential and the experimental.

But on the other hand, art cannot live only by means of revelation. Art must have, as shown us by the history of art of all times and all civilizations—indeed, it has an imperious need of—organization (including that of randomness), therefore of inference and of its confirmation, therefore of its experimental truth.

The two modes of inference and experimentation are today almost always closely related to the computer. Just as the wheel was once one of the greatest

products of human intelligence, a mechanism that allowed one to travel farther and faster with more luggage, so is the computer, which today aids the transformation of human ideas. Computers resolve logical problems by heuristic methods. But computers are not really responsible for the introduction of mathematics into music; rather, it is mathematics that makes use of the computer in composition. Yet if people's minds are in general ready to recognize the usefulness of geometry in the plastic arts (such as architecture and painting), they have only one more stream to cross to be able to conceive of using more abstract, nonvisual mathematics and machines as aids to musical composition, which is more abstract than the plastic arts.

Since World War II, computer science has invaded the domain of human activities. The arts, and in particular music, have not been overlooked by this tidal wave. Slowly in the 1950s, then accelerating, the computer and its peripherals have been spreading like mushrooms in the centers of musical activity, upsetting the attitudes of composers to a far greater extent than did the revolution of the tape recorder, which originated the first physically permanent memory of sound. The danger is great of letting oneself be trapped by the tools and of becoming stuck in the sands of a technology that has come like an intruder into the relatively calm waters of the thought in instrumental music. For we have already a long list of attempts at composition by the computer. But what is the musical quality of these attempts? It has to be acknowledged that the results from the point of view of aesthetics are meager and that the hope of an extraordinary aesthetic success based on extraordinary technology is a cruel deceit. Indeed, little of this music goes beyond the recent rich findings in instrumental music or even beyond the babblings of electronic music in the 1950s.

Why? In my opinion, the reasons for these failures are multiple, but we can single out two essential ones:

1. The musicians using computers are cripples in general theoretical ideas, especially in mathematics, physics, and acoustics. Their talent, whenever it exists, is powerless in penetrating the virgin domain where only abstract thought would be capable of guiding their experimental attempts, and it grasps but shadows.
2. The scientists having access to computer technology are sucked in by a sort of inferiority complex in front of the aesthetic aspect of music and, not having had to struggle on the aesthetic plane, are inexperienced and lacking and have no idea where they should be heading. Consequently, they fool around with mathematical and technical gadgets with the net musical result of very little, if any, artistic interest since they are not able, and do not know how, to employ talent when they have it.

In these two cases, artistic talent, as it can clearly be seen, plays—and must play—a determining role.

To escape from these impasses, the remedies are obvious: the first category of musicians should make an apprenticeship in the necessary sciences, and the second category should plunge into the delicate questions of talent and aesthetics, constantly experimenting with them by composing. But this will not suffice. It seems to me that the moment has come to attempt to penetrate more profoundly and at the same time more globally into the essence of music to find the forces subjacent to technology, scientific thought, and music.

I am going now to confine myself to sketching a single line of approach—since there are many—that appears to me to be very important. Indeed, research in the coming years must explore diverse levels, from microcomposition, which deals with sound synthesis starting from durations of the order of the microsecond (one millionth of a second), all the way to macrocomposition, which treats musical discourse for durations in terms of hours.

The methods and theoretical approaches may be distinct according to one level, or they may be used on more than one level. To throw more light on the problem, we are going to discuss two near-extreme levels: microcomposition and macrocomposition in the above-defined sense, by giving central ideas that serve as springboards for the coming years of research and composition aided by computers.

MACROCOMPOSITION

a. Explore compositions starting from the macroscopic ST(ochastic) program in FORTRAN (published in *Formalized Music*, 1971, Indiana University Press), which is stochastic in orientation and uses sound elements (1) of orchestral instruments, (2) designed on the UPIC[1], and (3) produced by the methods and theories of microcomposition (see description further on).

b. Explore the method of *polygonal variation*, the term we have given to a series of sound realizations. Roughly, this is a step-by-step construction of a pressure curve that is modified at each step (period) by a stochastic device, modified in the sense of time and also in the sense of the pressure values at each sample. Acoustical experiments using a computer and a digital-to-analog converter have shown that, for special values of the mathematical device, there arises a sort of probabilistic resonance that engenders rhythmic multiplicities of timbres, dynamics, attacks, and macroforms. The principal distributions

1. UPIC stands for Unité Polyagogique Informatique du CEMAMu. UPIC is a graphics-based computer sound-synthesis system at the Center d'Etudes de Mathématique et Automatique Musicales (CEMAMu) in Paris.

used until now are the *logistic* and *Cauchy* distributions (fig. 9.2). (See Appendix A.)

c. Explore a sort of "palindromization" with stochastically variable amplitudes, of the *polygonal variation* that makes possible a modulation of the preceding macroform on a higher level.

d. Explore clonings (in the etymological sense, arborescences) of polygonal variation; given a *polygonal variation*, a point stochastically chosen becomes a germ that engenders a new branch (a new polygonal variation) for which the characteristics are defined stochastically. This process may be applied to several *trunks* at the same time. (For realizations for instrumental music, see Appendix B.)

e. Explore Markovian processes on several interlocked levels. For example, we can consider clouds (configurations) of points, such as the Gabor grains (*Formalized Music*, p. 54) or grains that have been designed on the UPIC, and link them with the help of transition probability matrices in the discrete case or

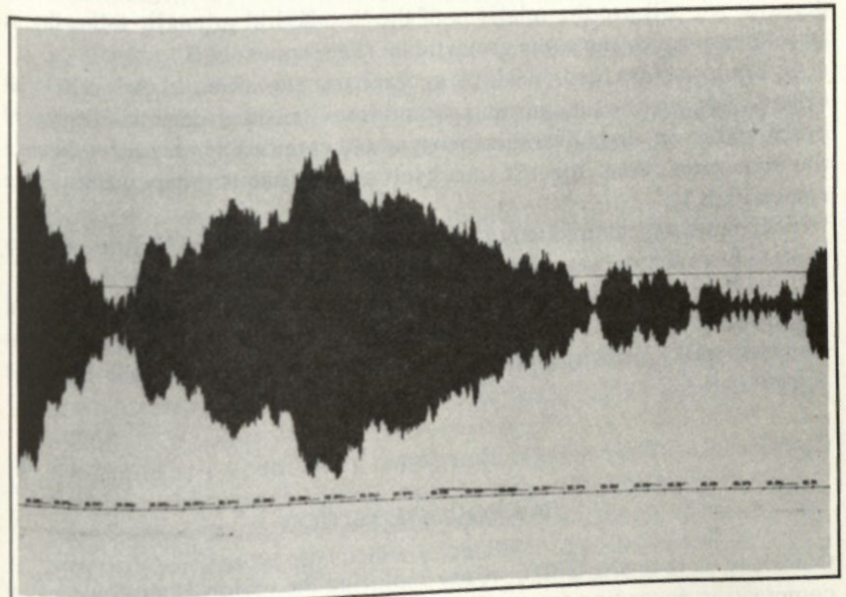


Fig. 9.2. Plot of "logistic" probability function summed and within special elastic barriers (random walk). Computer-generated sounds from this and other functions were used in the DIATOPE music.

with the z transform in the continuous case. We may then consider these linkings, in turn, as states and link these states by a Markovian process distinct in general from the preceding one. Therefore we need to explore chains that are nested, one inside the other. (See Appendix B.)

f. Explore Cartesian products of sets of points taken from the spaces of sound characteristics, using the structures of finite and infinite groups. Example: take a subset of clouds (configurations) of points (notes) or of designs made on the UPIC and consider the Cartesian product of these points with the points taken from a three-dimensional space (e.g., intensity, duration, density), but taking as a model the hexahedral group of the cube (subset of couples of Cartesian products provided with symmetries of transformations distinctive to the cube). This would occur in the outside-time domain. (By "outside time" I mean the domain in which time plays no role whatsoever. For example, the pitch intervals of the white piano keys form a structure that is in the outside-time domain. On the other hand, a melodic pattern based on the piano keys lies in the time domain, because their temporal order matters.) Other examples on a higher level of complexity are the structures of *Nomos Alpha* and *Nomos Gamma*, described in *Formalized Music*. For the inside-time domain, we will use the relations of the hexahedral group by following its structure, given by the cubic group table. (See Appendix B.)

g. Explore *sieve theory,* which generalizes the notion of the scale to all ordered sets, such as time instants, durations, intensities, densities, degrees of order, and so on, and inject this theory into the preceding areas of research in the first stage, then inject it into itself and/or use it independently. (See Appendix B.)

h. Explore logical functions applied to sets of sound characteristics or to sets of already structured sets. (See Appendix B: *Herma,* etc.)

i. Explore the generation of lines in any two-, three-, or n-dimensional sound space by defining each point as a function of probability functions (random walk, Brownian movement [*Formalized Music,* p. 246]). (See Appendix B.)

MICROCOMPOSITION

Outside of work on the UPIC, we are exploring the region of algebraic microcomposition according to non-Fourier methods, rather than Fourier-type methods such as Music V, to which most other laboratories limit themselves. This is what distinguishes our work at the CEMAMu. A statement of the problem follows.

Sound Synthesis Outside Fourier Synthesis (Analysis)

The central idea is based on the following two points:

1. A sound may be completely represented by its curve of atmospheric pressure variation in time. It is this curve that strikes our ears and nothing else. Consequently, to judiciously construct pressure-time curves (linear forms) goes back, in theory, to fabricating any desired sound through digital-to-analog conversion. This curve and its corresponding sound (music) will be considered as an *entity*.
2. The principle of repetition and of more or less faithful duplication is general and aids the comprehension of music at all its levels, from the microscopic to the macroscopic. On the microscopic level, for example, the ear not only detects faithful repetitions in the form of timbre but also takes into account their densities in the form of pitch. On the macroscopic level, canons, variations, and so on are equally immersed in this principle of renewal. Each event, wherever it occurs, is in a sense unique, separable, and not exactly reproducible because of the loss (even when it is almost negligible) in the fidelity of a possible duplication. This could be because of the time that has elapsed between two reproductions. But with a sufficient "approximation," they can "appear" identical (within the zone of approximation), forming *equivalence classes* in which the individuals are separable in general while merging in particular cases. The absence of repetition in the pressure-time curve is heard as noise, therefore as an extreme entity.

The dialectic union of these two basic points may be accomplished in three ways:

1. Starting from harmonic synthesis—that is, the strict periodicity of an elementary trigonometric form [$\sin(\omega t)$] produced by uniform circular movement and of its appropriate superpositions (Fourier)—one can construct, in theory, any more or less periodic waveform in the pressure-time space.
2. Starting from a deliberately nonperiodic waveform (Brownian movement) in the pressure-time space, one may proceed to inject periodicities—that is, duplications—either of fragments of the original wave or of sections set up separately, leading to a curve that is more or less periodic. We can see clearly the symmetry of these first two processes.
3. Starting from a pressure-time curve defined by some given function, be it probabilistic, algebraic, or trigonometric, one may continue by repeating this curve and at the same time injecting a stochastic modification into it after every repetition. This stochastic modification is chosen so as to

produce the statistically continuous negation of the original period, affecting the timbre, pitch, rhythm, intensity, and general evolution simultaneously. Now, in general, for any entity, let us suppose that the reproduction strays more and more from the entity of origin—in other words, that the deviation is applied at the same time to all parts of the entity. The entity will be pulverized into a statistical cloud of constituent elements. On the macroscopic level, we will have an amorphous cloud of sounds, rhythms, timbres, and dynamics; on the microscopic level we will obtain a Brownian curve that will be perceived as white noise. So we are introducing here the stochastic element as the limit of periodicity in the broad sense—in other words, renewal of the entity and at the same time a greater and greater negation in the reproductions.

At each reproduction of any entity, the entropy of the entity increases according to a certain *delta*—that is, the information describing the entity degrades partially at each renewal, irretrievably. It becomes the job of the composer to master, with intuition and reason at the same time, the doses of these entropy-deltas circulating through all the macro-micro-intermediate levels of the musical composition. In other words, one establishes an entire range between two poles—*determinism*, which corresponds to strict periodicity, and *indeterminism*, which corresponds to constant renewal—that is, periodicity in the large sense. This is the true keyboard of musical composition. Thus we emerge in a domain of multiple scientific and philosophic resonances. The continuity and discontinuity of the mathematicians and of the time-space of quantum physicists are such resonances.

The question that arises in all its generality is to know which mathematical construction to specify to the computer so that what is heard will be as interesting as possible—new and original. Without dwelling too long on this, I will cite an interesting example belonging to a case I was able to discover sometime ago by using the *logistic* probability distribution. For certain values of its parameters α and β and its elastic barriers, this distribution goes through a sort of stochastic resonance, through a most interesting statistical stability within the sound produced. In fact it is not a sound that is produced, but a whole music in macroscopic form. This form is the result of rhythmic transformations, giving a polyrhythm of events with variable timbre, changing pitches and intensities—in short, rhythmic strands of meeting and jostling sounds. I have used this procedure in the music of the *DIATOPE* at the Centre Pompidou (fig. 9.3).

To show to what extent this duality (that is, the entity and negation of the entity by varied reproductions at each step) is important, I put forward again and more explicitly the following question in the specific case of sound

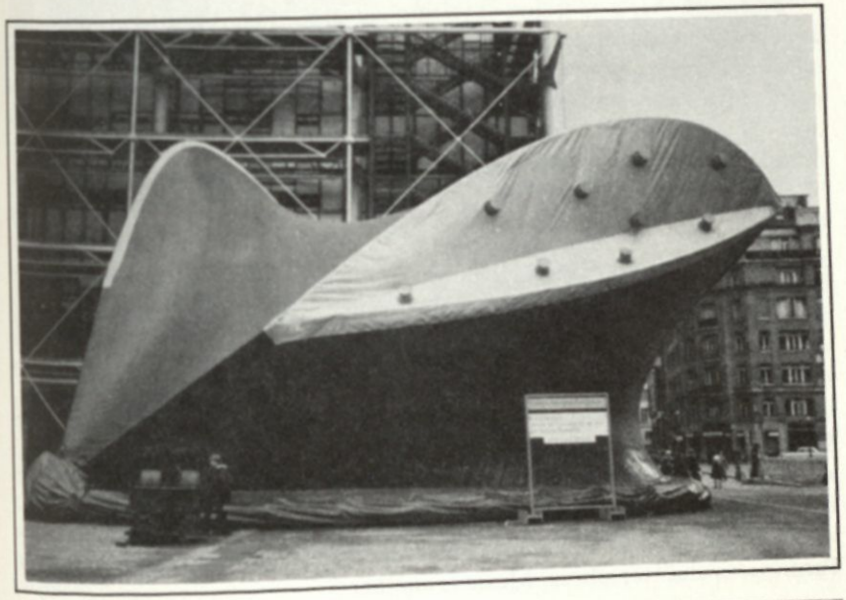


Fig. 9.3. The *DIATOPE*, rear view. Plastic dismountable construction in front of the Centre Georges Pompidou, Paris, 1978.

synthesis by computer and digital-to-analog converter: how can one obtain a rich, living, previously unheard-of sound? Does one start from an entity and its reproductions and inject probabilistic variations, creating greater and greater deviations from the initial entity, which tend toward a stronger negation? Or, on the contrary, should one start from an absolute negation—in other words a Brownian curve containing absolutely no germ whatsoever of an entity—and inject more or less varied reproductions of fragments of this curve, so as to engender progressively or explosively, an unheard-of, rich, living sound? In the first case, one would define the entity by strict periodic functions (trigonometric, for example) stacked or adroitly combined, then inject probabilistic perturbations at each reproduction of the entity. For the second case, one would define a set of functions of probability functions describing a specific Brownian movement that would constitute a furthermost negation. Then one would inject reproduction laws for connected or unconnected fragments of the Brownian curve to define the entity corresponding to these laws. These are two pathways, opposite and symmetrical, to rich, living, unheard-of sound. Naturally there is no exclusivity of one pathway over the other, and the results can be extremely interesting and strikingly different in the two paths.

Here is another expression of this universal duality, this time in philosophy, formed by the entity and its negation: the duality of the conflict opposing the thesis of Parmenides to that of Heraclitus. Parmenides decided that Being must exist always and everywhere, homogeneous without variation. Heraclitus decided that nothing is immutable, that everything changes. Thus expressed, these two positions are not compatible. They become compatible, however, if one decides that the Being of Parmenides is the entity that we invoke at the beginning. But it is an entity that would not last—as if time were formed of strings of cells and the entity inscribed in this bounded set of cells would not be able to avoid disappearance and death, once all the limits were reached in exchange for an imperfect reproduction. Then the perpetual change of Heraclitus is precisely realized by the reproduction of this entity in a chain of renewals—that is, in periodicity in the large sense. Thus, in this way, the Being of Parmenides conserves its integrity in the entity but is stained with temporal, spatial, and homegeneity limitations. Change, in general, cannot be instantaneous and total but is obtained progressively by a periodicity that is synonymous with varied reproduction, although it can be explosive at times. The universe of genetics is a beautiful and clear incarnation of this marriage between Parmenides and Heraclitus. Music is another.

COMPOSING WITH LIGHT

Composing with sounds for the ear leads us to compose with light for the eyes. The laser beam and the electronic flash are the equivalents of beautiful sounds. To make them gleam in space is to create music for the eyes—visual abstract music that would put galaxies, stars, and their transformation within the reach of humanity, on a terrestrial scale, of course (fig. 9.4). This music for the eyes is created with concepts and procedures stemming from musical composition. The result is a new art of vision and hearing that is neither ballet nor opera, but really an abstract spectacle in the sense of music, of the astral or terrestrial type. Movements of galaxies (sped up), storms, and aurora borealis are examples of what this new art not just recreates—this would be without interest—but truly creates with the means put at its disposal by the present technology.

Presently a new type of artist can master events of the size of a large city if given the means. And soon the artist will be able to go out into the cosmos. This is realized with and in the *DIATOPE.* I conceived and designed the *DIATOPE* in its plastic-fabric tent (a special architecture in hyperbolic paraboloids) for the inauguration of the Centre Georges Pompidou in Paris. Being itinerant, it could represent the Centre Pompidou in other cities in France or other countries. After Paris, it went in 1979 to Bonn, West Germany, where it was invited by the mayor of the city (fig. 9.5).

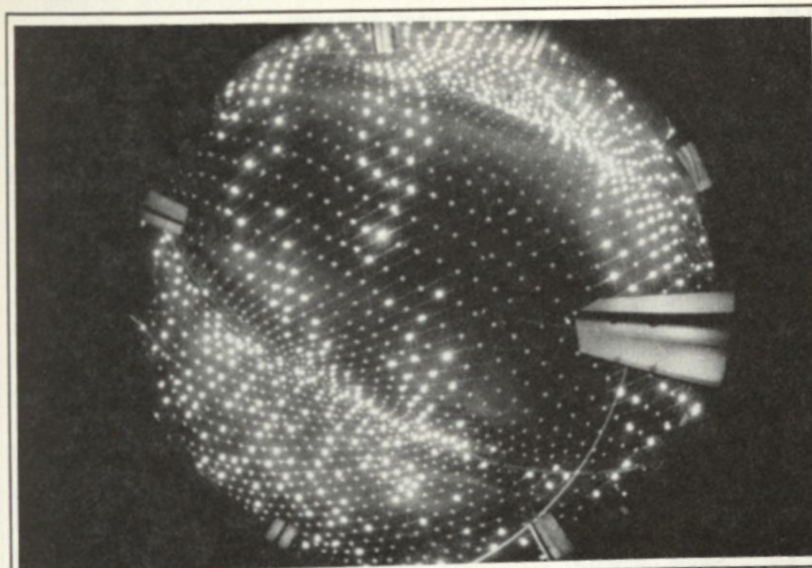


Fig. 9.4. Computer-generated flash patterns during the *DIATOPE* spectacle (fish-eye snapshot)

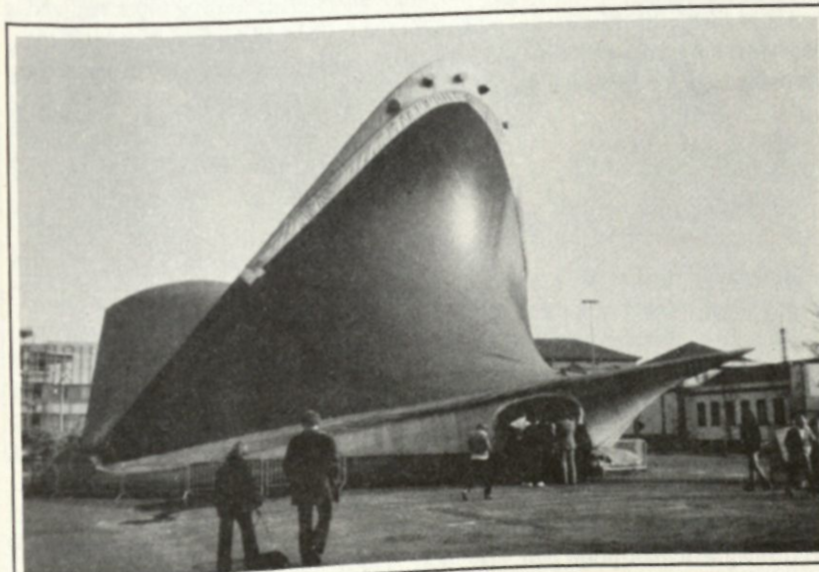


Fig. 9.5. The *DIATOPE,* front view and entrance, Bonn, 1979

In the *DIATOPE,* four laser beams (4 watts each) are equipped with optical devices that produce varied light effects. With these laser beams, 400 special mirrors create multiplicities of luminous spider webs in movement. Moving pools of light or sprays of luminous arrows trace in the space and on the black fabric of the tent trajectories of shooting stars or mosaics of bursts of light. Swirling configurations surround the spectator seated or lying on a glass-tile floor that lets through other events underneath (fig. 9.6). In addition, 1600 electronic flashes form revolving spirals, forms invading or disappearing in total blackness. These flashes are mounted on a metallic net suspended under the plastic shell. The music, recorded on seven tracks, is distributed automatically by the machine-program score (fig. 9.7), in continuous movements, to the 11 high-quality loudspeakers. The commands come from a nine-track digital tape drive that decodes an *image* of the set of simultaneous commands (around 2000), each twenty-fifth of a second. The commands are dispatched by cable to their destination in the space. The 46-minute spectacle consumes 140,500,000 binary commands. Naturally, to control and coordinate all these configurations, their transformations, and their movements, it is necessary to use the computer either interactively or by writing a digital tape according to a special light-machine program score. This digital tape, decoded each twenty-fifth of a second, commands the states of thousands of light sources or optical devices of this visual music. The light composition and the digital tape were realized at the CEMAMu; the music was realized at the CEMAMu and then completed at the electronic studio of the Westdeutscher Rundfunk (WDR) in Cologne, West Germany.

UPIC

To think music as composer, as craftsman, and as creator, it is first necessary to study solfège, notation, music theory, and even an instrument over a long time. And since, in addition, musical creation is considered superfluous, very few people are able to attain it. Thus the individual and the society are deprived of the formidable power of free imagination that musical composition offers them. We are able to tear down this iron curtain, thanks to the technology of computers and their peripherals. The system that has succeeded at this tour-de-force is the UPIC (Unité Polyagogique Informatique du CEMAMu). The principle is the following: on a special drawing board one traces designs with an electromagnetic ball-point pen. (fig. 9.8). These designs are read by the minicomputer to which the table is connected. The designs are interpreted, according to the choice of the user, as pressure curves, dynamic envelopes,

Fig. 9.6. View during the *DIATOPE* spectacle with the public on the translucent glass-tile floor

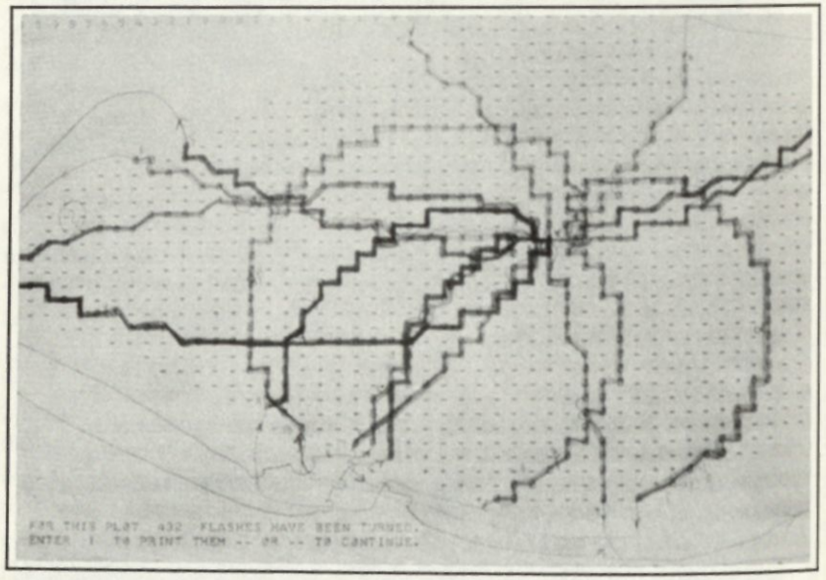


Fig. 9.7. Computer simulation of flash patterns for the *DIATOPE* spectacle (complex conformal transformations).

scores in the time-pitch domain, and so on. The computer calculates graphic command data, and the result, after being sent through a digital-to-analog converter, is heard immediately on the loudspeaker and recorded on a tape recorder or a digital tape drive. In this way one may create banks of waveforms, envelopes, and graphic scores. One may mix, delete, and realize many of the operations of a traditional electronic music studio by nothing more than pointing with the electromagnetic pen to various parts of the table that are sensitized like keys or buttons of an ordinary electronic device. Children may draw a fish or a house and listen to what they have made and correct it. They can learn, progressively through designing, to *think* musical composition without being tormented by solfège or by incomplete mastery of a musical instrument. But as they are led to construct rhythms, scales, and more complex things, they are also forced to combine arithmetic and geometric forms: music. From whence comes an interdisciplinary pedagogy through playing. All this is naturally valid for the "man on the street" and a fortiori for the researcher and the professional composer, since the sound is calculated in very thin slices of the thickness of 1/50 000 of a second.

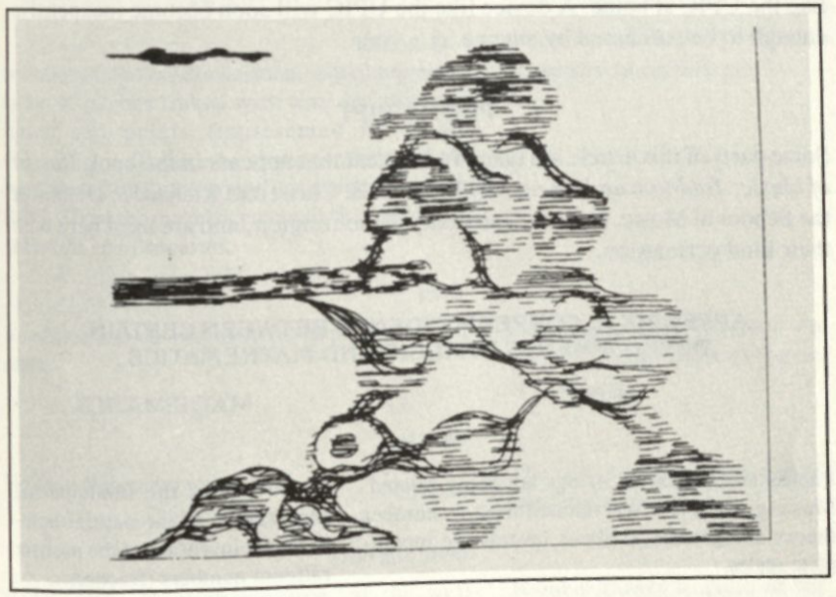


Fig. 9.8. Page from *Mycenae-Alpha* (1978) by Iannis Xenakis. First music composed on the UPIC computer/graphic system of the Centre d'Etudes de Mathématique et Automatique Musicales, Paris.

CONCLUSIONS

What emerges from all this is that for music and the visual arts of tomorrow it will be necessary to form artists in several disciplines at the same time, such as mathematics, acoustics, physics, computer science, electronics, and the theoretical history of music or the visual arts. They will need fundamental knowledge of a theory of forms and of their transformations, whether in paleontology, genetics, or astrophysics. It is important to encourage them and to give them the means to create with a system such as UPIC for musicians and an analogous one for visual artists.

But the touchstone of this evolution will lie in the training of a large number, of masses, as artist-creators right from the start of kindergarten all the way through the present national education in the same way as the massive training in the scientific disciplines in the high schools. To this end *telematics* will have a strong influence by making possible for the first time immediate creation at home through display terminals. It will enable distribution and public communication with feedback of individual realizations with the aid of a system like the UPIC at home. A device like the UPIC will soon become inexpensive enough to be purchased by anyone.

POSTSCRIPT

Some parts of this article are taken from a text that appears in the book *The Art of Music: Tradition and Change* by William B. Christ and Richard P. Delone of the School of Music, Indiana University, Bloomington, and are used here with their kind permission.

APPENDIX A: CORRESPONDENCE BETWEEN CERTAIN DEVELOPMENTS IN MUSIC AND MATHEMATICS

	MUSIC	MATHEMATICS
500 B.C.	Pitches and lengths of strings are being related. Music gives here a marvelous thrust to number theory and geometry. Music invents the incomplete scales.	Discovery of the fundamental importance of the natural numbers and invention of the positive rational numbers (fractions).
	No correspondence in music.	Positive irrational numbers—e.g., square root of 2 (Pythagorean theorem).
300 B.C.	Invention of the ascending, descending, and null intervals of pitches, in the additive language introduced by Aristoxenos, who also invents, in theory, a complete, equally tempered chromatic scale with the twelfth of a tone as the modulus (step). In parallel, there is a continuation of work with the multiplicative (geometric) language of the string lengths, which in fact is a translation of the additive pitch language (Euclid). Thus, music theory highlights the discovery of the isomorphism between the logarithms (musical intervals) and exponentials (string lengths) more than 15 centuries before their discovery in mathematics; also a premonition of group theory is suggested by Aristoxenos.	No reaction in mathematics. Number theory is left behind in respect to music theory and its practice, and it lies dormant in the west during more than 15 centuries in spite of the concept of infiniteness and of differential and integral calculus, first felt by Archimedes.

	MUSIC	MATHEMATICS
1000 A.D.	Invention of the two-dimensional spatial representation of pitches linked with time by means of staves and points (represented by Guido d'Arezzo), in advance by three centuries of the coordinates of Oresme and by seven centuries (1635–37) of the magnificent analytical geometry of Fermat and Descartes.	No parallel in mathematics.
1500	No response or development of the preceding concepts.	Zero and negative numbers are adopted. Construction of the set of rationals.
1600	No equivalence, no reaction.	The sets of real numbers and of logarithms are invented.
1700 and 1800	Rediscovery, through practice, of the well-tempered chromatic scale (acme with Johann Sebastian Bach). Music is now left behind in the field of basic structures. But, on the contrary, tonal structures, polyphony, and the invention of macroforms (fugue, sonata) are in advance of and bring to light the seeds that most certainly will inoculate a new life in the music of today and tomorrow. The fugue, for example, is an abstract automaton used two centuries before the birth of the science of automata. Also, there is an unconscious manipulation of finite groups (Klein group) in the four variations of a melodic line used in counterpoint.	Number theory is ahead of but has no equivalent yet in temporal structures. These structures will come later with stochastic processes, game theory, automata, etc. Invention of the field of complex numbers (Euler, Gauss), quaternions (Hamilton), the definition of continuity (Cauchy), and the invention of group structures (Galois, Abel).
1900	Liberation from the tonal yoke. First acceptance of the neutrality of the chromatic totality (Loquin [1895], Hauer, Schoenberg).	The infinite and transfinite numbers (Cantor). Peano axiomatics of the natural numbers. The beautiful measure theory (Lebesque, Borel, Heine).

Year	MUSIC	MATHEMATICS
1920	First radical formalization of macrostructures through the serial system of Schoenberg.	No new development of the number theory. A discussion of some older contradictions in set theory. (Music will catch up in the coming years.)
1930	Reintroduction of finer pitch gradations through the use of quarter tones, sixth tones, etc., although still immersed in the tonal system. (Vyschnegradsky, Haba, Carillo).	
1950	Second radical formalization of macrostructures with permutations, pitch modes of limited transpositions, and nonretrogradable rhythms (Messiaen).	
1953	Introduction of the continuous scale of pitches and time (use of real numbers) in calculating the characteristics of sound, even if, for reasons of perception and interpretation, the real numbers are approximated with rationals. (This is my own contribution, theoretical as well as musical, which included as well the use of various domains of mathematics such as probabilities, logic calculus, and several structures including group structure. These will play an important role later in macro- and microcomposition).	
1957	New formalizations in music on the macrostructure level: stochastic processes, Markov chains, though used in quite different ways (Hiller, Xenakis), and also the use of computers (Hiller).	
1960	Axiomatics of the musical scales with the *sieve theory* and introduction of complex numbers in composition (this is also a result of my work).	

Year	MUSIC	MATHEMATICS
1970	New proposals in the microstructure of sounds by the introduction of continuous discontinuity with the aid of probability laws (random walk, Brownian movement). This continuous discontinuity is extended to macrostructures, thus introducing another architectural aspect on a macrolevel—for example, in instrumental music (this also is a result of my work).	

APPENDIX B: MOSAIC OF COMPOSITIONS BY IANNIS XENAKIS

ST

ST/4-2
ST/10-080262
ST/48
Morsima-Amorsima
Atrées
Stratégie
Polytope of Cluny
(sound synthesis:
ST + Cos-G
Gabor signal)

Group

Akrata
Nomos Alpha
Nomos Gamma

Logical Operations on Classes

Herma
Eonta

Sieves

Persephassa
Nomos Alpha
Nomos Gamma
Mists
Nekuia
Aïs

Random Walks

Mikka
Mikka-S
Cendrées
N'Shima

Arborescences

Evryali
Erikhthon
Cendrées
Empreintes
Noomena
Phlegra
Khoäi

Polygonal Variations	Markov Chains	UPIC
Legend of Er (*DIATOPE*)	*Syrmos*	*Mycenae-Alpha*
Jonchaies	*Analogiques A & B*	*Anemoessa*
Ikhoor		

Spectacles of Light and Music

Polytope of Cluny: first fully automated spectacle of light
DIATOPE: fully automated spectacle of light and music, making use of all the other composition means on the macro-micro and the intermediate levels.

These two spectacles intertwined with other spectacles such as the *Polytopes of Montreal, Persepolis,* and *Mycenae.*

INDEX