CENTER FOR COMPUTER RESEARCH IN MUSIC AND ACOUSTICS

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Department of Music
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OVERVIEW
Center for Computer Research in Music and Acoustics
(Recent Work)

edited by

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CENTER FOR COMPUTER RESEARCH IN MUSIC AND ACOUSTICS
Department of Music, Stanford University
OVERVIEW

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INTRODUCTION

The Stanford Center for Computer Research in Music and Acoustics (CCRMA) is an interdisciplinary facility where composers and researchers work together using the computer-based technology as a new musical and artistic medium, and as a research tool.

Areas of ongoing interest at CCRMA include: Applications Hardware, Applications Software, Synthesis Techniques and Algorithms, Signal Processing, Digital Recording and Editing, Psychoacoustics and Musical Acoustics, Applied Pattern Recognition and Artificial Intelligence, Music Manuscript by Computer, Composition and Real-Time Applications with Small Systems.

The CCRMA community consists of administrative and technical staff, faculty, research associates, graduate research assistants, graduate and undergraduate students, visiting scholars and composers, and industrial associates. Major departments actively represented at CCRMA include music, electrical engineering, computer science, and psychology.

Center activities include academic courses, seminars, small interest group meetings, summer workshops and presentations. Concerts of computer music are presented each quarter with an annual outdoor computer music festival in July. In-house technical reports and recordings are available and public demonstrations of ongoing work at CCRMA are held monthly during the academic year.

Research results are published and presented at professional meetings, international conferences and in established journals including the Computer Music Journal, Journal of the Audio Engineering Society, and the Journal of the Acoustical Society of America. Compositions are presented in new music festivals and radio broadcasts throughout the world and have been recorded on cassette, LP, and compact disk.

Support for CCRMA has been received from the California Arts Council, the National Endowment for the Arts, the National Science Foundation, the Rockefeller Foundation (for artists-in-residence), the System Development Foundation, Doreen B. Townsend, Apple Computer, Dynacord, Sequential Circuits, Symbolics, Xerox Palo Alto Research, Yamaha, and private gifts.

Brief History of CCRMA

In 1964, while pursuing graduate studies with Professor Leland Smith, John Chowning began the work in computer music at Stanford using Music IV with help from Max Mathews of Bell Telephone Laboratories. Initial experiments were carried out with the help of the Computer Science Department on their time-sharing computer system (an IBM 1790 and a DEC PDP-1). Together, Chowning and computer science student David Poole put together the first on-line computer composition and synthesis system, with technical help from Computer Science and Electrical Engineering. As a result, John Chowning wrote the first programs for moving sound sources through a four-speaker space. As part of this project, a 12-bit DAC with a multiplex of four outputs was built. The program had both angular and distance cues, reverberation and Doppler shift.

In 1966, the Stanford Artificial Intelligence Laboratory moved to the D.C. Power Laboratory Building on Arastradero Road and acquired a DEC PDP-6 computer. Music 10, a music compiler for the PDP-6, was written by David Poole. It was at this same time that Chowning joined the music faculty teaching music theory and computer music, and the first course in computer–generated music was offered.

Exploratory work on musical timbres began in 1967 and led to the discovery of the use of frequency modulation (FM) for sound synthesis by John Chowning with the help of David Poole and engineering graduate student George Gucker .

In the summer of 1969 the first summer workshop in computer-generated music was taught by John Chowning, Leland Smith, Max Mathews and George Gucker.

The work in FM synthesis resulted in the composition "Sabelithe," written in the spring of 1971, and the publication of Chowning's paper on FM synthesis, which appeared in the Journal of the Audio Engineering

Society in September 1973. The FM synthesis technique was licensed to Nippon Gakki, Inc. (Yamaha) in 1974, and the development of electronic music instruments based on FM synthesis was begun in Japan with consultation from Stanford. A patent for FM was granted in 1977. Other basic work by Leland Smith included the development of SCORE, a computer program written in FORTRAN which enabled composers to synthesize and compose pieces using the DEC PDP-6 and later the PDP-10; and MS, a music manuscripting program which has been highly developed over the years and is now available for use on personal computers.

Working on dissertations in the area of computer music, James A. Moorer, Loren Rush, John Grey, and F. Richard Moore made important contributions to the field in analysis-synthesis, digital editing, and synthesis hardware. In 1972, John Grey and Andy Moorer began working on the analysis and resynthesis of real instrument tones by computer. This work led to important discoveries about the psychoacoustics and perception of sound. In 1968 Loren Rush began working in the area of digital recording using programs written for speech research. Working with Andy Moorer and Ken Shoemake, in 1974 he completed EDSND, a program for computer editing of recorded sound, and in 1976 the first high quality digital recordings were made.

Early compositions from CCRMA included: "Sabelith I" for sound and 3 performers by John Chowning in 1966 (never completed due to the Artificial Intelligence Laboratories move to the DC Power Laboratory Building); "Adosaman" for tape, by Irmfried Radauer in 1967; "Rondino" for stereo tape, by Leland Smith in 1968; "Pour" for sound and recorded voice, by Martin Bresnick in 1969; "Fragment" for stereo tape, by Martin Bresnick in 1970; "Sabelithe II" for quad tape, by John Chowning; "Machines of Loving Grace" for bassoon and narrator with stereo tape and "Rhapsody for Flute and Computer" for flute and stereo tape, by Leland Smith in 1971; "Turenas" for quad tape, by John Chowning in 1972; "A Little Traveling Music" for amplified piano and quad tape, by Loren Rush in 1974; a realization of John Erickson's "Loops," by John Grey in 1974; and "Song and Dance" for orchestra and quad tape, by Loren Rush and commissioned by the San Francisco Symphony in 1975.

Because of their growing reputation, members of the computer music group at Stanford were asked by Pierre Boulez in 1973 to participate in the planning stages of his music research institute being formed as part of the Centre Pompidou in Paris. In August 1975, the IRCAM group came to Stanford to participate in a special workshop on computer music. The research relationship and exchange between the two centers has continued over the years.

In June of 1975, CCRMA was formed with funding provided jointly by Stanford University, by a gift from Mrs. Doreen B. Townsend, by a grant from the National Science Foundation for research, and by a grant from the National Endowment for the Arts for computing equipment for musical purposes. As a result, CCRMA was able to commission the design and fabrication of the Systems Concepts Digital Synthesizer (designed by Pete Samson and called the Samson Box) which was installed at CCRMA in 1977. Although a part of the music department at Stanford, CCRMA continued to share facilities and computing equipment with the Stanford Artificial Intelligence Laboratory (SAIL) of the Computer Science Department at the D.C. Power Laboratory Building on Arastradero Road. The founding co-directors of CCRMA were faculty members John Chowning and Leland Smith and research associates John Grey, James A. Moorer and Loren Rush.

Funded research at CCRMA at this time included work on "Timbre Perception for Complex, Time-Variant Tones" and the "Computer Simulation of Music Instrument Tones in Reverberant Space."

The first computer music concert ("An Evening of Computer Music and Film") was held August 10, 1976 at Dinkelspiel Auditorium and in 1978 CCRMA presented a concert of computer music at the Stanford Museum of Art.

Additional work accomplished at this time included the development of software for the Samson Box. The initial program was MBOX written by graduate student Gareth Loy in 1977-78 and resulting in his piece "Nekyia" in 1979. Subsequently graduate students, David Jaffe, Michael McNabb and Bill Schottstaedt revised and extended this software into a program called SAMBOX as a result of their own compositional work. In 1977 Marc LeBrun began work on waveshaping synthesis techniques. In 1978 Bill Schottstaedt began work on Pla, an interactive interpreter program which includes a graphics-oriented note-list editor.

This program, written in SAIL, was first used in the composition of Schottstaedt's "Daily Life Among the Phrygians" and has become the main program used by composers at CCRMA for compositions that use the Samson Box digital synthesizer.

Another major accomplishment in 1977 occurred when graduate student Michael McNabb, using available software for the PDP-10 computer and additive synthesis, digitally applied the timbres of vocal sounds to instrumental sounds to achieve a smooth transition in timbre between the two. This work resulted in the composition of "Dreamsong" in 1979. This work also led to experiments by John Chowning applying this technique to FM synthesis (in Paris in 1979), which resulted in the composition of "Phone" in 1981, which in turn led to Stephen McAdams dissertation work on spectral fusion.

This was a compositionally active period of time. Works written at this time include: "Dirge" and "Sinfonia for Computer" by Bill Schottstaedt and "Stria" by John Chowning in 1977; "Sandcastle," "Mars Music," "New Music Liberation Army," "The Gong Tormented Sea" and "You're So Far Away" by Bill Schottstaedt, "Mars in 3D" by Michael McNabb and Bill Schottstaedt, "Oracle - 4am" by Paul Wieneke and "Standing Waves" by Stuart Dempster in 1978; and "Nekyia" by Gareth Loy, "Dreamsong" by Michael McNabb, "the servent snapping eye..." by Roger Reynolds, and "Daily Life Among the Phrygians" by Bill Schottstaedt in 1979.

In November 1979, the Artificial Intelligence Laboratory moved to the Stanford campus to new facilities with the Computer Science Department. CCRMA remained at the D.C. Power Building and, with the help of Stanford University and Yamaha, obtained its own time-sharing computer system: A Foonly F2 (later upgraded to a Foonly F4) central processor with 256K of memory emulating a DEC PDP-10 and designed by David Poole, various computer peripherals, and several digital/audio workspaces. Extensive work was accomplished by Andy Moorer and Tovar in writing software for the new system. This included a comprehensive signal processing library for computer music applications by Andy Moorer. No longer having to share computers and work space with the Computer Science Department enabled CCRMA to become an independent and fully functioning center.

In 1980, CCRMA received a grant from the National Science Foundation (NSF) to begin work in the "Intelligent Analysis of Acoustic Signals." Initial work was begun by Loren Rush and Chris Chafe at CCRMA in conjunction with Joseph Rockmore and Bernard Mont-Reynaud at Systems Control Technology in Palo Alto. This work has continued at Stanford under the direction of Bernard Mont-Reynaud and with support from NSF.

In 1982, CCRMA received a major five year grant from the System Development Foundation for operating support and research. This grant enabled the center to obtain needed equipment and support staff. The Center was able to accommodate a larger number of composers, researchers, and students and computer music at Stanford began to flourish.

Important work at this time included dissertations by Stephen McAdams, John Strawn, Jeff Borish, Christopher Sheeline, John Gordon, and Andy Schloss in areas of psychoacoustics, and Julius Smith on digital filters and physical modeling.

Work in the area of digital recording continued under the direction of Loren Rush. Software and hardware interfaces were extended to enable the direct digital transfer of sound between the Foonly and Sony PCM-F1 and PCM-1610 digital recorders. This work resulted in the mastering of "The Digital Domain", a compact disk demonstrating the capabilities of digital audio and released by Electra for Warner Special Products in January 1984. Other recording projects completed included "Michael McNabb: Computer Music", a digitially mastered phonograph disc reseased in 1984 and "Computer Music from CCRMA" produced by Janis Mattox.

In 1982 Julius Smith and David Jaffe began work on the synthesis of plucked strings using the Karpus-Strong plucked-string algorithm. This resulted in the composition of "Silicon Valley Breakdown" by David Jaffe in 1983 and has led to the exploration of other synthesis techniques based on physical modeling by Chris Chafe, David Jaffe, and Julius Smith.

Other compositions during this period included: "Towers of Hanoi" by Andrew Schloss and "Garden for

Orpheus" by Paul Wieneke in 1980; "Phone" by John Chowning, "Attend" by Paul Wieneke, and "Voicespace IV" by Roger Reynolds in 1981; "Colony" by Bill Schottstaedt from 1981-1983 and "Book of the Burning Mirror" and "Dinosaur Music" by Bill Schottstaedt in 1983; "Diptych"for stereo tape, mezzo-soprarno and string quartet by Jonathan Berger, "gamelan R gong gong" by JoAnne Carey, "Mr. Normal and the Details" and "Red Cup and Rat (What's Wrong With This Picture?)" by Doug Fulton, "Bristlecone Concerto No. 2" for solo violin, solo mandolin, instruments and tape, and "Bristlecone Concerto No. 3" for mandolin, percussion and stereo tape, by David Jaffe, "Music for S" by Stanislaw Krupowicz, "Dialogos" by Servio Marin, "Shaman" for percussionist, dancer, bassist, vocalist, and tape by Janis Mattox, "Getz Variations" for tenor sax and tape by Dexter Morrill, "Anira" by Adolfo Nuñez, "Daybreak" by Bill Schottstaedt, "Pentateuch" for soprano, three choral groups, large orchestra and tape by William Sussman, "Areyto" for chamber orchestra and sound by Raymond Torres-Santos, and "Etude (Hommage a Bartok No. 2)" by Amnon Wolman in 1984.

In 1983, the university began to make plans to move CCRMA to the Stanford campus. With the completion of the Music Department's Braun Music Center, the former home of the Music Department, the Knoll, became available for CCRMA's use. (Built in 1916, the Knoll was originally the home of Ray Lyman Wilbur, president of the University. The Wilbur family lived in the building until the early 1940's when the building began to be used by the University for academic purposes.)

Renovation of the Knoll to provide CCRMA with facilities for interdisciplinary digital acoustic research and composition began in April of 1985. CCRMA moved to the Doreen B. Townsend Center for Computer Research in Music and Acoustics in the reburbished Knoll building at the end of March 1986.

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contributed by Patte Wood, ***Administrative Director***

CCRMA ROSTER

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Staff & Faculty	
JC	John Chowning, Director
PAT	Patte Wood, Administrative Director
HMK	Heidi Kugler, Secretary
MVM	Max Mathews, Professor (Research)
LCS	Leland Smith, Professor
JRP	John Pierce, Visiting Professor Emeritus
EDS	Earl Schubert, Professor Emeritus
BMR	Bernard Mont-Reynaud, Senior Research Associate
CC	Chris Chafe, Technical Coordinator/Research Associate/Lecturer
BIL	Bill Schottstaedt, Research Associate
JOS	Julius Smith, Research Associate/Lecturer
EG	Emmanuel Gresset, Research Assistant
TVR	Tovar, Programmer/Analyst
JAY	Jay Kadis, Audio Engineer

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DK	Douglas Keislar	PhD Music
STK	Stanislaw Krupowicz	DMA Music (Composition)
DKM	David Mellinger	PhD CS
XJS	Xavier Serra	PhD Music
BOB	Bob Shannon	PhD EE

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			Project
ECB	Emily Brant	DMA Music (Voice)	Computer Music Seminar
	Michael Cohen	PhD Computer Science	MIDI Seminar
EZC	Estrella de la Cruz	DMA Music (Composition)	Composition
GRD	Glen Diener	PhD Music	Research
YIE	Young-Iee Eom	Masters Music(Composition)	Computer Music Seminar
JON	Jonathan Franklin	PhD VTSS	Computer Music Seminar
HMH	Martin Herman	Music/U.C. Berkeley	Computer Music Seminar
HBH	Harlan Hokin	DMA Music (Voice)	Research
	Brent Johnson	PhD EE	Signal Processing Seminar
RSK	Richard Karpen	DMA Music (Composition)	Composition
LEH	Ben Knapp	PhD EE	Research
TOS	Steven Lakatos	PhD Music	Research
CCL	Chris Lanz	DMA Music (Conducting)	MIDI Seminar
LFT	Alfred Leung	PhD Music	Research
	Todd Mozer	GSB	MIDI Seminar
DCO	Chris Overton	Math	Computer Music Seminar
DVO	Daniel Oppenheim	DMA Music (Composition)	Composition
DAV	David Perry	PhD Psychology	Research
AMI	Ami Radunskaya	PhD Math	Research
TKA	Alex Tkaczevski	DMA Music (Composition)	Computer Music Seminar
CYW	Cheng Wang	DMA Music (Composition)	Composition
XSK	Xu Sika	Masters Music (Composition)	Computer Music Seminar
DDZ	David Zicarelli	PhD Hearing and Speech	Research

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	Robert Armas	Undeclared	MIDI Seminar
	Joseph Belfiore	Undeclared	MIDI Seminar
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	Gregory Cohen	Undeclared	MIDI Seminar
PVC	Peter Commons	EE/Music	Computer Music Seminar
DJC	Daniel Culbert	CS/work study	MIDI Seminar
SCD	Scott Douglas	EE	Computer Music Seminar
SBF	Steven Fram	Philosophy	Computer Music Seminar
TFG	Tim Gallagher	Undeclared	MIDI Seminar
	Peter Habicht	Undeclared	MIDI Seminar
	David Hornik	Undeclared	MIDI Seminar
	Martha Horst	Music,VTSS	Computer Music Seminar
DEB	Deborah Jue	Undeclared	Signal Processing Seminar
ABS	Andrew Leary	Music/work study	Computer Music Seminar
	Jeffrey Neal	Undeclared	MIDI Seminar
	Eric Ranelletti	Undeclared	
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	Carl Wescott	Undeclared	MIDI Seminar
	Timothy Westergren	PolySci	MIDI Seminar
CPW	Chanel Wheeler	Computer Science	Computer Music Seminar

Visiting Scholars/Composers

JWB	James Beauchamp	Professor; University of Illinois (sabbatical Jan-June)
BRG	Jonathan Berger	Assistant Prof. Music; Yale University (sabbatical 87)
AB	Al Bregman	Professor of Psychology, McGill University (sabbatical 86-87)
	Diana Deutsch	Professor of Psychology; UC San Diego 1988 (spring 88)
PEF	Pablo Furman	Visiting Professor,Music; UC Berkeley 1988
GEG	Guy Garnett	Visiting scholar
JEG	Johannes Goebel	Composer; German government fellowship (1987-88)
JH	Jonathan Hallstrom	Professor of Music, Colby College, Rockefeller fellowship
DAJ	David Jaffe	Composer
FLM	Fred Malouf	Composer
JRM	Janis Mattox	Composer; NEA fellowship
MMM	Mike McNabb	Composer
	Simon Millward	Composer; English fellowship (1987)
DEX	Dexter Morrill	Prof. of Music; Colgate University NEA fellowship
IJM	Ira Mowitz	Composer; Guggenheim fellowship (Sept. 87 for 1 year)
ARP	Arnaud Petit	Composer; French fellowship (March 88 for 1 year)
MUZ	Loren Rush	Composer
TAK	Teiichi Takenaka	Prof. of Music; Japanese fellowship (Sept. 87 for 1 year)
LEO	Leonello Tarabella	Italian fellowship
HKT	Rick Taube	Composer; Rockefeller fellowship
PAW	Pascal Willequet	Visiting scholar; French Government/Renault fellowship
AYW	Amnon Wolman	Composer; Lecturer, UC Berkeley

RESEARCH

Introduction

Computer music is inherently an interdisciplinary field. Accordingly, the research mentioned in this section spans such areas as engineering, computer science, and psychology, as applied to various musical topics. Some of the papers include substantial work in all these fields. It should be added that much effort, not necessarily reported here, continues to go into general software development and the creation of suitable environments for composition and research. This section does not pretend to be a comprehensive description of the research being done at CCRMA. For a more complete description a list of recent publications is included in section 9 of this report and a complete list of CCRMA research publications is available upon request.

Although most of the research is interdisciplinary, the following rough categorization of the articles presented here may be useful. Engineering research at CCRMA includes both hardware development (Mathews) and various aspects of digital signal processing (Cook, Garnett, Serra, Shannon, Smith). One active area is physical modelling of musical sounds (Chafe, Cook, Garnett, Smith). Research in computer science includes representations for music (Diener). Various types of work have been done in psychoacoustics and neuropsychology (Keislar, Lo, Perry, Pierce). The NSF project in machine perception (Mont-Reynaud and Gresset) couples the work of a computer scientist and that of an engineer. The researchers contributing to this section include graduate students, faculty, staff, and guests.

Simulating Performance on a Stringed Instrument

Chris Chafe
Research Associate

I've been involved with a continuing project in which music is synthesized from a physical model controlled by simulated physical performance gestures. This report presents some issues in designing a control system for such a synthesizer with illustrations of a running system. The present system runs considerably slower than real-time and can be viewed as an enviroment for answering questions about live control of an eventual real-time system. The kinds of control in a playable real-time instrument are similar to those used here, where the computer interprets the musical score.

The present method is specific to synthesis from a physical model of a bowed string. The simulated performances are used to generate five time-varying control signals for the model. Its externally controlled parameters are: string length, string damping and bow speed, pressure and position. Other instrument types could be tested using a similar approach. The set of parameters and articulation rules would be adapted to fit their synthesis models.

Time-varying envelopes for the parameters are created by concatenation of short envelope segments corresponding to performance gestures. Gestures themselves are not represented in actual physical terms, so there is no notion of the extent or rate of motion of the fingers or bow. Instead, the system represents gestures in terms of their effect on the string parameters.

Musical scores are coded as lists of gestures. The method is a variant of tablature notation, in which a score is described from the point of view of hands manipulating the instrument, rather than pitches on a stave. For example, events described in this fashion include ***martele attack on string III*** in the right hand or ***hammered pitch at the fifth on string I*** in the left hand.

When creating a performance, the system adds details that cause gestures to conform to the changing state of the musical phrase. The intended result is consistency in the complex gestures that perform multiple notes as well as some amount of expressiveness in the rendition.

It should be mentioned that though computer music synthesis using physical models is promising, it is still largely a theoretical field. Commonly restricted to running in software, sound generation is many times slower than real-time. Real time systems require either a new type of synthesizer hardware or general purpose super computers. The bowed string algorithm used in the present work runs in software. Since the purpose here is to discuss control of the cello model, background concerning the algorithm itself should be found by referring to the following sources: Cremer, McIntyre and Woodhouse, Smith and Weinreich.

A physical model of an acoustic source can represent a vibrating system of arbitrary complexity. A complete cello-like synthesizer is constructed by coupling together a number of bowed string models that simulate each string's internal reverberation and the bow's frictional driving mechanism, and which are further coupled to the bridge, the sound box and the air. Weinreich has termed the bowed string algorithm a method for "synthesis from first principles." It is a simplified, but nonetheless accurate, description of a dynamic physical system. The algorithms representing the four cello strings are iterated in time from some initial state, eg. an open string at rest. As the bow excites string motion, waves begin to circulate that are emitted as sound at the model's output. Events in the musical score, through the intermediary control system, cause changes to parameters controlling the bow and string.

A cellist develops skill at manipulation of five basic parameters. ***String length,*** controlled by fingers on the player's left hand, determines the round-trip time of the recirculating waves and the resulting pitch of the sound. The right hand controls ***bow speed, pressure and contact position,*** affecting loudness and tone quality. A fifth control, ***string damping,*** controls the amount of wave recirculation and varies with movement of the fingers and bow touching the string.

In synthesis experiments with a physical model of the cello, a number of advantages have become clear:

- Regimes of oscillation are correct in the time domain. Self-sustained oscillators such as bowed strings are ***dynamical systems.*** The system's ***attractor*** (equilibrium state) under normal bowing conditions is a waveform exhibiting ***Helmholtz motion,*** named after the acoustician who first described the periodic mechanism of string capture and release by the bow.

- As in the real instrument, transient behavior of the system is state-dependent. Response to a particular articulation depends on what the system was doing in the recent past.

- The model has the same external controls as the physical system. These are the same parameters a cellist develops skill at manipulating.

- There is an "intuitive feel" to the system's response. A cellist can recognize and imitate a synthetic articulation and recommend improvements in the control values. The complete range of cello articulations can be synthesized.

The most important restriction concerning control of the model, is that the model's state should persist through successive articulations and events, allowing the system to produce real-sounding transients. Traditional speech or music synthesis techniques, eg. linear predictive coding, formant synthesis, additive synthesis, frequency modulation or sampling methods, don't easily produce real-sounding transients. In physical models, finely detailed transients are achieved when new events interact with the reverberation of previous events. A particular articulation, played twice, may sound different as it interferes with a system that is already in motion. The state of the system at any given point in time is a result of complex interactions between external control parameters and recent system "memory." Due to this accurate transient behavior, dynamical system models are useful in a wide range of real-world simulations [Campbell].

This accuracy creates, in music synthesis, recognizable instruments. A wide range of acoustic cues contribute to the dynamic signature of an instrument in play, especially features that are co-varying. It doesn't matter particularly ***what*** is being performed. For instance, the model can produce tones or passages that are recognizably the sort produced on the instrument by unskilled players. Refining the performance of the control system is reminiscent of early exercises on the instrument. When something is amiss, the best approach has been to compensate the controls as if listening to the actual instrument. Tone quality, for example, might be improved by specifying something like, "...use more bow pressure at that point."

Music is performed on the cello synthesizer by a control system that replaces the cellist. It is gesture-based, and imitates the effect of the player's actions on the strings. The system also attempts to reproduce some of the interpretive functions of the player such as musical phrasing. Time-varying envelopes for the synthesis parameters are calculated by rules that correspond to basic gesture segments. It takes a few of these to form the complete envelope of a musical note.

The system operates from musical scores coded in common notation and enhanced with explicit markings for fingering (which string) and bowing. These markings are often added by players to their parts, indicating fingering and bowing choices for performance of a particular passage. Each succeeding pitch is associated with either a fingering change, bow change, or both. The old forms of tablature style scores for lute are reminiscent of this approach, in that they notated the time-varying placement of the hands on the strings.

The gestures that the hands can perform vary in complexity. The simplest items are those that initiate a bow stroke or finger a new pitch. More intricate operations are possible which result in coordinated activity across multiple strings, such as rolled (broken) chords. The system breaks each gesture into strings of smooth sequences of short envelope segments, for example, those resulting in string release, bow accleration, pitch sustain or vibrato. The larger gestures described by the musical notation, such as a note sequence, trill or slur, are formed by compounds of these "atomic" segments.

For each gesture in the score, sequences of segments are cued up in a time ordered list by the control system. In most cases this sequence contains one or more transition segments paired with a sustain. For example, a change of bow is comprised of a quick deacceleration, a re-attack and a sustain segment.

The extendable duration of sustain segments distinguishes them from transitions, whose durations are fixed. Sustainable segments prolong the string state, possibly modulating it as with vibrato and tremolo.

The intervening transitions determine trajectories for the hands as they move between sustainable motions. For example, left hand transitionals include sliding, hammering, or homing in on a new pitch.

Each segment contributes to the envelopes of a few synthesis parameters. The number of parameters that will be inflected synchoronously depend on the particular segment. In a reversal of bow direction, the re-attack segment affects four parameters: bow speed, pressure, pitch and damping.

Several kinds of phrasing marks are likely to be encountered in a score and are referred to as ***phrase controls***. These include dynamic level, vibrato rate, tone quality, and tempo, among others. Separate slow changing envelopes which are internal to the system are generated from marks in the score. Their levels represent the ***phrase state*** at any given instant.

Phrase controls are updated by an ever-present background process. These values are relatively stable since they change over the course of seconds or tens of seconds. Though a given control level usually persists longer than a single note, exceptions can occur, for instance the ***messa di voce*** which is a dynamic swell within a prolonged tone.

The control system can insert unscored segments when interpreting a score. Some may belong to a particular playing style while others arise from a need to simulate a natural level of "sloppiness." Citing some inaccuracies common in string playing: pitches are rarely placed directly on target by the left hand and spurious sounds often accompany position shifts or string crossings. Computer-perfect synthesis without a dose of nature's noises has a lifeless quality.

Novice players are familiar with the burden of tracking the many controls that they're responsible for: the bow, intonation, tempo, loudness, and timbre. In effect, this burden of control can be diagrammed as a "polyphonic" texture, containing several independent "voices" for which the hands are the articulators.

From the instrument's point of view, the two articulators that act on it are sometimes coordinated and sometimes completely independent. Both hands can simultaneously contribute to the value of a synthesis parameter. This is clearly the case for the damping coefficient in the string, less obviously so for string length and the bow parameters.

Two independent processes represent the hands in the control system. These processes will be assigned different kinds of gestures that invoke string excitation, and pitch or damping change. Regarding string excitation, besides the right hand bowing the string, the left hand can hammer down a new pitch and either can pluck. Given the possibility for left hand sounds, the current system allows the left hand process to add excitation by inflecting the bow parameters. A hammered pitch is simulated by a sharp jerk in the bow envelopes.

Updates to the synthesis parameters occur every 10 milliseconds. Prior to each update, the hand processes evaluate their contribution to the control envelopes. Their respective contributions are then summed for each of the synthesis parameters.

Segments in the two hands may be changed synchronously or asynchronously depending on the type of musical material. In typical performance, this texture may switch rapidly from a one-to-one correspondence, to many-to-one or entirely independent relationships between the hands. An additional parallel-executing process evaluates the state of the phrase controls. The hand processes evaluate their segments within this context. The desired effect of this mechanism is that a given gesture or articulation will translate into different envelope shapes depending on the dynamic level or tone quality.

Control processes, synthesis processes, gestures and segments are objects in the system's computer program. Each has an associated set of execution rules. At initialization time, the instrument configuration is declared giving the number of strings and their tunings. The system then copies the required number of each object prototype. From 1 to 4 strings are played by 2 hands (additional hands can be made available for more difficult scores).

All processes execute each tick in an order that follows a hierarchical pattern. The instrument is the ***father process*** and hand, phrase and string synthesis processes are offspring. The father contains a scheduler that cues the time-ordered segments into its daughter control processes. The evaluation order (at each update) is as follows: instrument, phrase controls, and then string by string: left hand control and right hand control. The updated value is sent to the string synthesizer process which then outputs 10 milliseconds of sound.

The system has been used to synthesize some examples from a small "taxonomy" of bowed articulations. Five features are taken into account in distinguishing articulations:

- Reversal of Bow Direction. This distinguishes ***detache*** from ***legato*** bow strokes.
- Initial displacement. Significant bow pressure before the bow starts to move causes a hard attack. The initial string snap that results is also called a ***quasi-pizzicato.*** Examples are ***martele, marcato and staccato.***
- Pulsed accent. Soft attacks have no discernable displacement, but have accentuation after the string is in motion with ***lance, porte and loure*** bowing.
- Duration of separation. Tones separated by different (brief) amounts of silence include ***martele, spiccato and staccato.***
- Off-string bowing. Lifted or ricocheted strokes contrast with completely on-string tones. During the decay segment, off-string tones ring where their on-string counterparts are damped. ***Pique, spiccato, jete and flying staccato*** strokes are played off-string and are distinuished by amount of bow bounce.

The control system has been used in several different musical situations, both with hand entered scores and scores composed by algorithmic means. In experimenting with different instruments, envelope-generating rules have been created that simulate guitar and koto performance. The rule base can be expanded for a selection of musical styles, instruments or even the personal touch of different performers.

The rules currently in the system have been refined through repeated synthesis trials, in the absence of measurement from actual playing. In the future, such measurement is possible through instruments which can sense bowing and pitch information. An example is the system of Askenfeldt's. Under development at CCRMA are string family MIDI controllers which will allow analysis of live performance, and which will help to determine more precise envelope-generating rules.

Note lists, which are the most common data structure for specifying computer music scores, would be inappropriate for synthesis from physical models of continuous sounding instruments like the cello. Control of the instrument involves several parallel processes whose synchrony varies depending on the musical texture. Slurs are a simple example of asynchrony between the two hands of a cellist. Subsequent notes continue the sound before the previous note is extinguished. In the note list style of control, only separate and isolated notes can be sounded and slurs are approximated by keyboard style note overlap. Moreover, the state of the synthesizer is re-initialized each note, eliminating the advantages of physical modelling for synthesis of natural-sounding transients.

Recapping the issues that have been considered in the design of the cello control system: Left hand and right hand gestures are merged in a way that permits their effects to interact in the synthesis. Gesture segments can coordinate changes in several parameters at once and track evolving phrase conditions. The synthesizer's state is preserved allowing the physical model to be put to full advantage.

The author wishes to thank Julius Smith, Xavier Rodet and Gabriel Weinreich for contributions of advice and software. The system was first implemented in 1984 at IRCAM Paris, France and again in 1987 at CCRMA.

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Implementation of Arbitrary Bore Shapes Using Digital Waveguides, with a Specific Realization of the Saxophone Conical Bore

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Research is being done in the accurate physical modeling of wind instruments, specifically the conical bore and the lip-mouthpiece forcing function. The desired end result is a set of economical algorithms for computer simulation and synthesis. Waveguides are being used as the fundamental modeling blocks of the physical vibration systems. It is expected that waveguide techniques will yield the necessary complexity of musical characteristics, while providing simplicity of computer implementation.

Object oriented programming techniques in Smalltalk-80 were used as the environment for initial simulations. A working model of the waveguide clarinet (Julius Smith) was completed and tested. From this model, the extension was made to the soprano saxophone. Measurements were taken from an actual instrument, and a waveguide model of the bore was built using one scattering junction per spatial sample. The non-linear reed table was retained from the clarinet model, and with little difficulty and fine tuning, a working saxophone was realized.

Work was continued on the saxophone bore, with the goal in mind of reducing the computational complexity of the algorithms. The bore was successively decimated (approximated with fewer and longer cylindrical segments), and various characteristics were measured. With the measured comparisons of the various levels of approximation, and the corresponding computational requirement data, a selection could easily be made as to how to realize the model on particular hardware.

With the general-bore object used for the saxophone, many interesting departures are possible. Among the possible directions for this research are the implementation of time-varying bores, which have applications in speech and singing voice synthesis.

The Ariel DSP-16 signal processing board is also being investigated as a device for simulations, since a logical direction to pursue for the clarinet and saxophone would be a real-time implementation with provisions for real-time control inputs. This would allow a saxophone player to be linked to the program, so that measurements and further fine tuning could be done.

Removal of Reververation from Prerecorded Musical Signals Using Adaptive Filters

Perry R. Cook

A study of reverberation removal by means of adaptive filters was undertaken in the spring of 1987, and continues. The algorithms take advantage of the uncorrelated nature of reverberation at different locations within a listening area. Since the adaptive filter adapts to cancel any part of the filtered signal which is uncorrelated with the desired signal, the diffuse reverberation is cancelled.

Tests were performed on a variety of signals, first a synthetic signal and simulated reverberation, then a real signal with digital reverberation, and finally real signals with real reverberation. Reductions of up to 20dB were achieved. It was found that the more stationary the source signal (sustained tones), the more the process itself became audible in the resulting filtered signal. Research continues in heuristics which could reduce the audibility of the adaptive process itself, while increasing the amount of reverberation removed in sections where it is most pronounced.

Forest:
A Hierarchical Score Representation Scheme

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Advances in computer hardware are continually creating new possibilities for the composer of electro-acoustic music. With every emerging technology, a corresponding need for software to access that technology is created. Unfortunately, software designers have all too often been content with reimplementing the techniques of previous technologies. Though the perpetuation of such time–honored techniques as note lists and unit generators is not without merit, the opportunity to strike out in new directions should not be missed. In my view, the design of computer music software is much more than simply an exercise in software engineering: it is an integral part of the compositional process itself. Like composition, it is an expression of how the designer feels about one aspect of the process of music making. Accordingly, my research is focused on the exploration of novel approaches to the design of such software.

At CCRMA, I have been investigating an original scheme for the organization of computer music scores. This scheme, an outgrowth of previous research in grammatically based compositional processes, takes the form of an active data structure for the temporal organization of basic musical objects into compositions. The scheme makes no assumptions whatever about what these objects might be or how they behave, hence it is capable of organizing a variety of different synthesis technologies in the same score. Further, since the structure is basically a tree graph, it shows similarities to many hierarchical views of musical form so prevalent in contemporary music theory.

At present, the scheme is implemented as a set of Smalltalk-80 classes collectively known as "Forest." Forest objects currently exist for both software synthesis and real-time MIDI control. The Smalltalk environment, with its powerful graphics capabilities, makes the design of a visually-oriented programming interface for the manipulation of Forest objects an enticing possibility—the design of such an interface is currently under development.

Waveguide Piano Project

Guy E. Garnett
Visiting scholar

Since January 1987, a project has been under development to model the physics of the acoustic piano efficiently and accurately. This is part of a larger project that seeks to model at least one of each family of standard orchestral instruments using what is known as "waveguide digital filters."

The piano, however, proves to be a particularly difficult system to model. One of the principal factors contributing to this difficulty is the complexity of the basic structure of pianos. They can be thought of as three subsystems, each one complex in itself, that are combined together to interact in still more intricate ways. These three subsystems are the strings, the soundboard and the pedal. Each of these contributes its own effect to the overall sound of the piano. Research to date has been successful at modeling the strings and the special attack characteristics imparted by the hammer. It should be a simple matter to extrapolate this to account for the effects of the pedal but, because of the sheer number of strings that would thus have to be simulated, this has not proved to be computationally practical. Instead, a system has been devised for summing the effects of a small number of strings and modifying this with other filters to approximate many strings.

One of the remaining unsolved problems is in modeling the soundboard. Because of its complicated geometry and the variety of forces under which it must operate, the impulse response of the soundboard—and hence its basic sounding characteristic—is difficult to model directly. Since part of the goal is to make a computationally efficient model, one that could actually be used to play music, it is not possible to use the most straightforward implementations of the model. So, instead, the first idea was to model only the gross features of the soundboard and hope that these would come close enough for most purposes. This has not proved possible, however, thus current research aims at coming to a fuller understanding of the way the soundboard resonates so that a new model can be developed that will better approximate it without too much added expense.

Several preliminary piano models have been developed using the Systems Concept Digital Synthesizer (the "Samson Box"), and a prototype has also been developed for a Lisp Machine 2 from Symbolics, equipped with an array processor, the FPS120b from Floating Point Systems. Work is currently underway to implement more general modeling tools for use on single-user workstations. It is my belief that greater progress in modeling specific systems will come about once an environment has been developed that will support a more flexible interactive approach. To this end, I am planning and testing software tools that will work effectively with the newest generation of signal-processing chips, such as the Motorola 56000, and new high powered personal computers such as the Macintosh II from Apple.

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Perception of Nonstandard Tunings

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The use of nonstandard tuning systems in music composition is an area that has remained largely unexplored in the past. Nearly all composers have limited themselves to the standard tuning, which for a couple of centuries has been equal temperament. The two most likely reasons for this restriction are the difficulty of learning to play in nonstandard tunings on traditional instruments, and the unavailability of more appropriate instruments on which composers could try out novel tunings. Both these factors are eliminated in computer music.

Because of the traditional restriction to standard equal temperament, listeners have "overlearned" the intervals of this tuning system. To understand more about the possibilities of arbitrary tunings, it is helpful to learn about listeners' perception of intonation, particularly the intonation of nonstandard intervals. A series of pilot studies were carried out to examine several questions in this area. The third of these is being elaborated upon as a dissertation in Music. Since the results described below are based only on pilot studies, they should be considered tentative.

1. ***Estimation of the sizes of intervals.***
   Subjects heard two complex tones sounded simultaneously and attempted to estimate the exact size of the interval, either directly in cents or by marking the corresponding point on a line graph. One subject also heard sinusoids ("pure" tones). The task was found to be rather difficult; however, the following observations were made:
   a. Nonstandard intervals tended to be judged as being somewhat closer in size to the nearest standard interval than they were in reality. This tendency was observed in the subjects who performed with the most accuracy, and is probably an influence of categorical perception.
   b. Often the approximate amount of mistuning from the standard interval was judged correctly while the direction of mistuning was mistaken.
   c. The intervals of just intonation were identified with high accuracy by subjects who had knowledge of their sizes from music theory.
   d. As might be expected, the task is easier if all the intervals share a common bottom pitch (rather than undergoing random transposition), and also when complex rather than pure tones are used.

2. ***Identification and learning of interval tunings.***
   The single subject was told a number in cents representing the "target" interval, which in most cases was 430 cents (a nonstandard interval). Two harmonic intervals close to each other in size were played, and the subject had to identify which one was the target. An adaptive and interactive paradigm was used to facilitate learning of the target interval. Results:
   a. Typical asymptotes for identification of the 430-cent interval were 8-10 cents.
   b. Asymptotes for a familiar consonant interval, the (just) perfect fifth, were around 2-4 cents. For a dissonant interval, the tritone, they were more like those of the nonstandard 430-cent interval.
   c. Identification was easier with simulated musical instrument sounds than with sine waves ("pure" tones).
   d. When this study was followed by a repeat of pilot study 1 (above), no significant improvement was found at the 430-cent interval. Categorical perception effects seemed to be lessened, however.

3. ***Effect of beats on perceived intonation.***
   According to psychoacoustic models of consonance, the presence of beating partials affects the subjective

purity of a harmonic interval. This study examines the perceptual effect of varying the beat rate independently of the fundamental frequency ratio. The goal is to determine the extent to which the percept of "out-of-tuneness" is governed by the actual interval size as opposed to the acoustical epiphenomenon of rapid beats. Some initial observations drawn from listening to synthesized examples are:

a. Coincidence of partials (i.e. lack of beats) reduces the out-of-tune percept in radically mistuned major triads.

b. A "just" triad with added beating sounds less pure than an equal-tempered triad with the beats removed. However, without beats, the just triad sounds more pure or "fused," which implies that the beats of nearly coinciding partials are not the only cue.

c. Beat rate, which is a function of register, is important for the out-of-tune percept; the same intervallic structure has different degrees of "out-of-tuneness" in different octaves.

This study is being continued in greater depth, using tones derived from recordings of musical instruments.

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A Treatment of Timbre—Progress Report

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A system, equivalent to variable-rate block-sampling in conjunction with (time-domain) transformations, was developed for the analysis/synthesis of natural timbres in conformance with their dynamic character. A linearly weighted metric perceptual importance tree was developed as a model of timbre perception with the aim

1. to form a bridge between psychoacoustic observations and the huge dimensionality of the control parameter space for the synthesis of timbres;
2. to facilitate the interpolation of timbres of diverse characteristics;
3. to facilitate data reduction with arbitrarily prescribed fidelity.

A coherent theory of timbre was developed which provides an absolute formal definition of timbre suitable for rigorous analysis and synthesis of timbre, and which includes Helmholtz's theory as a limiting case. The treatment, like Helmholtz's, is ***response***-based, rather than purely ***signal***- or ***stimulus***-based, such as the various existing transform methods.

Minsky's ***Society*** idea together with the notion of redundancy was applied to provide a model for feature-extraction from the single-input multiple-output response pattern in the cochlea. As a result, data compression on the engineering level is directly derived from data compression on the cognitive level—as it should be done. For details see (Lo, 1987).

Currently, Lo is developing the system on a Macintosh-like computer equipped with 16-bit D/A & A/D converters and 44.1 kHz sampling such as the Dyaxis board from IMS, San Carlos, California.

He has developed a second generation of his kinematic synthesis program in the fall of 1987. The program has been used for his research on and composition with timbres since.

His current concerns include the possibility of systematically using timbre, as opposed to pitch, as primary material in composition; the transferability of structures and hierarchies from one perceptual dimension to another; the form of local timbre space; the issue of timbre interpolability and the articulation of timbre relationship in terms of the internal structure of timbres. He is also searching for a systematic description of the dynamics of acoustic transients using the Poincaré Phase-Space approach.

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Pickups for the Vibrations of Violin and Guitar strings Using Piezoelectric Bimorphic Bender Elements

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A new method for picking up the vibrations of the strings of violin family instruments has been designed and constructed. The technique uses piezoelectric bimorphic bender elements to sense the vibrations. They can be made both very small and very sensitive. It is possible to install these sensors in the bridges of traditional instruments or to install them in special electronic instruments. By cutting appropriate slots in the bridge, the various strings can be decoupled and the output of each string sensed separately. Normal types of strings can be used and can be strung on the instrument in a traditional way, thus making it easier for traditional instrumentalists to use the pickups.

A bridge has been constructed for the electronic cello built by Chris Chafe. The instrument was played in a demonstration concert. The signals not only had an excellent timbre when amplified and sent directly to a loudspeaker (with a little added reverberation), but also the signals reliably actuated a pitch follower which could then be used as a MIDI control.

Obtaining a satisfactory pickup for the string vibrations in the electronic string instruments (violin family instruments and guitars) is difficult. The pickup should faithfully reproduce the vibrations of the string as an electronic voltage. In order to do this the pickup should have the following properties:

1. The pickup should be located close to the string so no intervening structure modifies or filters the vibration.

2. The pickup should be properly oriented with respect to the principal plane of the string motion.

3. The string motion should be closely coupled to the pickup.

4. The pickup should be part of the bridge of the instrument, since that is where the vibrations are transmitted to the body in an acoustic instrument.

5. The pickup should allow the string to be mounted in a normal fashion, i.e. to pass over a bridge with a small notch in it so the player will be able to replace the string easily and quickly.

Most commercial pickups are deficient in one or more of these properties, and this partly accounts for the big difference in timbre between electronic and acoustic instruments. Electromagnetic pickups used in standard electric guitars are never mounted in the bridge. Piezoelectric pressure-sensitive pickups have been mounted in violin bridges, but they are either mounted far from the string or in an orientation in which coupling to the string is marginal or in a way which makes replacing the string awkward.

All commercial piezo-electric pickups with which I am familiar use pressure-sensitive elements. It is difficult to properly couple the string vibrations to such an element. If the string passes over a notch in the bridge and the pickup is embedded in the bridge in any orientation, then the coupling will be unsatisfactory because the sound velocity in the bridge is so high that transmitted string vibration will cause the entire pickup to vibrate rather than putting vibratory pressure on the pickup. If the pickup is placed under one foot of the bridge, it is so far from the string that the bridge structure will seriously filter the vibration— usually it will attenuate the important high-frequency components.

One method for properly coupling the string to the pickup is to have the string press against the actual pickup itself in an orientation so that the string vibrations are normal to the pressure-sensitive surface of the pickup. This method does give a good waveform from the pickup. However, in order to have the string press against the pickup the string must bend slightly where it crosses the pickup so that the tension in the string will press it against the pickup. Arranging the bridge and the tailpiece to produce such a bend makes it awkward to mount the string on the instrument and causes some problems in tuning the string.

The essence of the device which I am describing is the use of a bender bimorph piezoelectric element as a pickup instead of a pressure-sensitive piezoelectric pickup. The structure of a bimorph consists of two layers of oppositely-polarized piezo material with a metallic layer between the two piezo layers and metallic electrodes on the outer surfaces of the piezo layers. When a force is applied to bend the pickup, one of the piezo layers will be in compression and the other will be in tension. Since the layers are oppositely polarized, the voltages produced in the piezo layers will add and the sum of the voltages will appear across the electrodes. In practice, the device is very sensitive, and a small deflection will produce a substantial voltage—a response which typically may be several volts.

When using the bimorphic pickup in a violin, the bridge is essentially normal except that the usual perforations are omitted. These perforations tune the bridge and cause it to couple to the violin body, both functions being unnecessary in an electronic instrument. The violin string passes over a notch in the bridge in a normal manner. Two slits are cut into the bridge on either side of the notch so that the string is actually supported by a narrow beam of material which is free to bend from side to side. The primary direction of string vibration is in a plane perpendicular to the axis of this beam, so the string vibration couples well into bending the beam. The piezo bender element is inset into the center of the beam so that it is bent along with the beam by the string motion.

The piezo bender elements are so sensitive that excellent signals can be obtained from very small elements. Thus it is possible to insert elements into a normal-sized violin bridge and to use this bridge on an acoustic violin. In this way one can achieve an amplified acoustic violin as well as an electronic violin with these elements.

The waveshape of the vibrations of a normal violin string is well known and is a sawtooth function of time. If a pickup produces a faithful sawtooth wave, then it is generally considered to be a good pickup. The waveshape from the piezo bender is a good sawtooth waveform.

One other property of the proposed structure is that it decouples the outputs of the individual strings from each other. Since each string has its own pickup and since each string is supported by a separate beam, vibrations from one string produce very little response in the outputs from the pickups on the other strings. Having the strings decoupled is absolutely necessary when an electronic violin is used as a control sensor for a synthesizer, since it is often desired to use each string to control a different aspect of the synthesizer sound. Having the strings decoupled is also desirable for an amplified or an electronic violin since it enables the player to individually adjust the loudness and timbre of each string and thus to get an instrument which has a uniform sound on all strings.

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Pattern Recognition in Sound, Gesture and Image

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1. Overview

This project has a long history at CCRMA. Early work was supported by two 2-year National Science Foundation grants under the title "Intelligent Systems for the Analysis of Digitized Acoustic Signals" [1,2,3,4,5,6,7]. Support from the System Development Foundation allowed the research to continue. A new proposal was written[8], and NSF support resumed in May 1987. Support for the project also came (unexpectedly) from the French Government, which provided part of the support for one project member, in the form of a scholarship.

The research focuses on the fundamental problem of separating multiple sources in a monaural acoustic signal. The approach relies both on the refinement of earlier ideas, and on the introduction of new ones, from psychoacoustics, computer vision and man-machine interaction research. Ideas from psychoacoustics and psychology of perception, notably the work of A. Bregman and S. McAdams [11,12] on streaming, which has deep roots in a Gestalt approach to the psychology of vision, have not yet made their way into the implementation. But several components of a visually-based approach are present already.

2. Signal processing front end

Much recent work has aimed at expanding our front end signal processing techniques that we have already found useful in earlier studies. The signal processing front end that we have developed includes fast yet powerful tools for visualizing frequency domain representations of sound. It allows interactive event detection and labeling. This front-end system is intended as a basis for further research in polyphonic sound detection, segregation and identification, but it also represents an experiment in man-machine interactivity.

The approach to spectral analysis embedded in our front end is based on the simultaneous use of multiple sampling rates. Fast Fourier transforms are performed in quasi-parallel manner on progressively decimated versions of the original sound signal. From the analyses performed at the original (highest) sampling rate one obtains the highest possible time resolution. Decimated signals yield poorer time resolution but higher frequency resolution, which is a better tradeoff for the lower frequencies. While the usual approach to time vs. frequency resolution tradeoffs implies that a choice must be made among several alternative views, we have developed techniques that integrate the information from multiple views.

The efficiency of our implementation partly comes from splicing the frequency representation of a musical signal into octaves—a musically meaningful operation which corresponds to doubling or halving the frequency scale. Octave decimation, which is an undersampling by a factor of 2, is implemented as a half-band lowpass filter (performed as a simple pointwise multiply in the frequency domain) followed by a time decimator of a factor of 2 (throwing away every other time sample).

In order to permit real-time processing, which will be possible with the use of DSP chips, the concepts necessary to an on-the-fly implementation have been developed. They include data-driven sequencing of events—blocks are processed as soon as they are available—and the careful choice of data structures.

The interactive display environment allows users to create, examine and annotate spectrogram-like data. A user can see the view or "plane" resulting from a single time/frequency resolution tradeoffs. Alternatively, he or she may combine multiple planes into a single view. This tool has already been very effective for manually exploring issues of acoustic segmentation, which we need to understand thoroughly to develop the next generation of automated segmentation tools.

Several methods for combining views have been found useful, and others could still be added to the system. The first combination method we designed was initially seen as an efficient means to approximate the data obtained from a constant-Q filter, i.e. a bank of bandpass filters where the width Δf of a frequency bin is proportional to the center frequency f, so that $Q = \Delta f/f$ is indeed a constant. The "Bounded-Q" algorithm gets its efficiency from that of the FFT. It approximates Constant-Q to within a ratio of 2 by collecting FFT results at the highest available frequency resolution. In other words, one keeps only the upper octave of each FFT block at each decimation. These octaves are combined on a single display using a $log(f)$ scale for frequency, so that each octave has the same height, and each bin has essentially constant width ($\Delta log(f) = \Delta f/f$).

A combination method we designed later attempts to combine the highest frequency and time resolutions available in the system on a single display. The frequency resolution used here is the same as in the previous method, which was optimized for frequency resolution at the expense of time resolution. Here, most of the highest time resolution is contained in the first computed plane, that is, the one that results from the original, non-decimated signal.

The views are combined by a surprisingly simple operation, point-wise min, an operation which can be implemented very efficiently through the use of bitmap operators, in combination with "texture half-toning". The resulting display does appear to offer both the highest time and frequency resolution. We are still working on the theory underlying this technique [paper in preparation] but we have already found it to be of great value in practice.

3. Image processing in pitch extraction and event detection

The two-dimensional representation of sound that we find most useful for general screening, and for some processing as well, is related to both conventional spectrograms and musical scores, in that time increases from left to right, and frequency increases from bottom to top. We use physical time on the x-axis, and the logarithm of frequency on the vertical axis. As a result of the front-end signal processing discussed earlier, the resolution achieved is fixed on both axes (Δt and $\Delta log(f) = \Delta f/f$). Direct experience with such a display representation indicates that it may present a good general compromise for perceptual mapping sound to image. One is then tempted to address event detection and pitch tracking by visual methods on this kind of display.

We now discuss one aspect that has been explored for starters. In simple harmonic sounds, the pattern of equidistant harmonics $\{k * F\}$ or $\{k\} * F$ on the linear axis becomes an equally recognizable pattern on the log axis, $\{log(K * F)\}$ or $\{log(k)\} + log(F)$. In the latter case, however, the vertical arrangement we call "harmonic comb", $\{log(k)\}$ (scaled to the display resolution) sweeps all possible fundamentals without changing shape or size. Or to put it another way, image convolution of our spectrogram-like representation with the harmonic comb acts as a simple pitch detector which operates at all times and frequencies in parallel.

We are not suggesting here that "harmonic comb convolution" is a plausible model of pitch detection in the human ear, or even a good pitch detection algorithm per se. But a round of investigations of pitch detection has begun, in which image processing will play an important role. Other problems in event detection and identification, including the detection of attacks and the recognition of "common fate" may also be approached by means of image processing, and the preliminary steps we have taken in that direction appear very promising, in comparison with more traditional approaches.

4. Musical analysis

Even though the current funding emphasizes analysis at the acoustic level, not the analysis of higher-level musical structures, the latter have received some attention in the early part of 1987. Progress included developments in the Xerox Lisp Machine implementation of ANA—a flexible musical performance analysis system capable of transcribing melodies with minimal user intervention. These melodies may come from standard MIDI recording data, from pitch tracking data, or from the results of analysis of acoustic recordings. Two recent additions to the system are worth noting.

First, ANA uses an algorithm for hierarchical clustering of performed durations, and offers the option of displaying the formation of clusters. In practice, an improvement of great value has been to add the option of editing the resulting clusters, by interacting directly with the graphic display of clusters, using the mouse to merge or un-merge clusters at will. From the man-machine interface point of view, this is an interesting form of cooperation. The system uses a fairly elaborate algorithm to come up with a guess at clusters, but manages to present its results in a visually editable form—one that actually tunes naturally into the human user's ability too "see" clusters effortlessly. The "manual override" option offered by the cluster editor allows user intervention before subsequent processing, during which durations are quantized to simple relative musical durations—a crucial step towards obtaining a conventional music score. The quantizer itself cannot yet receive advice from the user, but it usually behaves well when the clusters are correct.

The second major addition to ANA is described as a separate section, because it was largely conceived and realized as a separate effort, even though part of it "lives happily" in the ANA environment.

5. *Real Time Tempo Tracking*

This work was done in cooperation with Roger Dannenberg of Carnegie-Mellon University. As part of his research on real-time accompaniment techniques, which included a method to track the performance of a score, Roger became interested in moving from the harder problem of tracking a player during an improvisation, and chose the traditional blues as a style to experiment with.

The correlation technique Roger had devised for tracking position within the 12-bar grid needed the assistance of a technique for real-time tempo tracking. This had always seemed like a logical evolution for ANA's non-real-time kind of tempo analysis. ANA offered the best environment to carry out the research in, and to design a real-time algorithm. But the real-time implementation was done in C, on a PC, in the MIDI toolkit. The paper was presented at ICMC[9], complete with live demo (Roger drew much applause with his trumpet).

6. *New explorations: Optical Scanning of Musical Scores*

Over the years, a gradual realization has come that the project was not about Music Transcription from Sound, as originally presented, but about two complementary endeavors: a scientific one concerned with intelligent signal processing, pattern recognition or machine perception, and a musical one concerned with creating bridges between musical representations [10].

Musical performances can be represented as sound (a digital recording) or as gesture (a MIDI recording), whereas musical scores can be represented as abstract text, with some underlying structure (score languages) or as a visual object (printed score). While music transcription goes from sound to gesture to abstraction to image, musical performance (when reading a score) goes from image to abstraction to gesture to sound. In both cases, the "output" paths are the most traveled; much computer music work has been concerned with going from abstractions to sound or to printed scores. But the input paths, those involving machine perception, provide less traveled paths that this project tends to delight in.

In any case, for these reasons or for sheer curiosity, initial investigations have been performed to study the feasibility of optical scanning of musical scores, and the suitability of this problem as a research goal. The results were very encouraging. Projects in which music is read from paper could lead to exciting questions and new applications. Opportunities exist for doing something new and meaningful. Rather than exhausting possibilities, the Tsukuba robot [13,14] has showed an example of what is possible. Some interesting and useful material was gathered, and contacts were established with other researchers interested in this topic, notably Henry Baird at Bell Labs. All this may get assembled into a paper and a proposal eventually.

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Hemispheric Specialization for the Perception, Immediate Recall and Reproduction of Melodies

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In a melodic reproduction task, subjects (experienced pianists) listen to a short melody (7-13 notes), then immediately attempt to reproduce it with their right hand on a keyboard synthesizer. The accuracy of melody reproduction after hearing melodies in the right ear is compared to accuracy after hearing melodies in the left ear. Since the contralateral ear-hemisphere connection is stronger (or a greater number of cortical neurons fire in response to the contralateral ear signal), it can be inferred that significantly better performance with one ear reflects some degree of specialization for melody perception of the opposite hemisphere.

Conflicting results have been found with studies of melodic recognition, a type of task involving a yes/no or same/different verbal response after hearing a melody. In particular, apparent shifts to a right ear/left hemisphere advantage have been noted when experienced musicians are the subjects. It has been suggested (Bever and Chiarello, 1974), and demonstrated by comparing performance to subject reports (Peretz and Morais, 1980,1987) that task strategy may be the factor that leads to this ear shift, e.g. using an analytic strategy leads to a left hemisphere advantage, for both musicians and nonmusicians.

Considerable evidence from clinical studies following hemispherectomy, brain damage, hemisphere anaesthetization, neuroimaging, etc. suggests that melodic perception is more right hemisphere lateralized. However, demonstrating consistent, reliable left ear advantages for melody perception has proved to be much more elusive than demonstrating right ear advantages for language perception. Although musical processing is almost certainly less lateralized than language processing (Marin, 1982), part of the reason may be that melodic recognition tasks add a left-hemisphere or bilateral-frontal bias to the task, at least for subjects inclined by experience to analytic processing. This hypothesis is supported by preliminary evidence from BEAM (brain electrical activity mapping) studies during a musical cognitive task (Seashore Rhythm Test). Brain electrical activity at first appears to be concentrated over posterior zones, while the subject is listening to the rhythmic pattern of tones, then shifts to frontal zones as the subject prepares to respond (Duffy et al, 1985), perhaps while he is carrying out the cognitive activity neccessary to verbally answer the task question. In contrast, melodic reproduction may involve more principally the posterior sensorimotor zones.

Both "re-cognition" and "re-production" imply a re-processing after melodic perception, which is necessary to produce behavioral output. Recognition implies a cognitive-verbal reprocessing which may involve entirely different neural substrates than the initial melodic perception. Reproduction implies a cognitive-motor reprocessing, which, while also involving a broader functional-anatomical substrate, can be presumed to involve the more closely related substrates of acoustic-motor integration. Use of a non-verbal, motoric task output is a step toward further specifying the functional anatomical substrates of melodic perceptual skills.

It is expected that the results will indicate a stronger overall left ear advantage across subjects, because the task does not require the use of language. If task strategies identified from subject reports (e.g. analytic vs. non-analytic or reflexive) correlate with ear advantage (e.g. REA vs LEA), then further support will have been found for the hypothesis that shifts to a right ear advantage for melodic perception are associated with a more analytic processing mode.

In conjunction with this study, comparisons are also being made between the ear advantage for melody perception and the results of the octave illusion (Deutsch, 1974), which correlates strongly with handedness. Most right-handers hear the high tone in the right ear (although it is actually being presented alternately to both ears). The underlying basis for this correlation is not known, although of course most right-handers also process speech preferentially in the left hemisphere. There may be no correlation between ear advantage for melody perception and the high frequency in the octave illusion, or there may be a strong correlation, such that those right-handers with an atypical ear advantage for melody perception will also have atypical results for the octave illusion.

One reason that little study has been done of ear differences for melodic reproduction is that available methods for recording and measuring the accuracy of melodic reproduction using tape recording were tedious and unreliable (Kimura, 1967). I am employing MIDI recording of keyboard performance, which greatly facilitates recording and analysis of the large number of reproduction trials needed to investigate statistical ear differences. An Apple Macintosh microcomputer is used for data recording and analysis, using MIDI files produced by the Opcode MIDIMAC sequencer, and converted to a note list in spreadsheet form including pitch, duration, and velocity for each MIDI event. (Recording resolution is on the order of +/- 1.25 ms).

Preliminary investigations are also in progress of piano performance recorded via MIDI, through the analysis and graphing of agogic and dynamic phrases. The question of whether such patterns correlate with particular ear advantages will be considered, as well as relative synchronicity or independence of the right and left hands, but this type of performance analysis is also of interest in its own right, as an indicator of the motor programming which underlies expressive, skilled musical performance (Shaffer, 1981).

One further question is whether hemispheric specialization is changed by years of experience and high-level skill aquisition (or is found to be different in individuals with a high level of musical aptitude). Current plans include comparison of ear advantages and task strategies between musicians of varying levels of experience. Does the consequent decrease in the extent to which low-level aspects of skill must be consciously attended to lead to greater or lesser evidence of hemispheric specialization? Is this shift associated neuroanatomically with a lesser role for frontal areas and a greater role for posterior sensorimotor and association areas, or with greater cerebellar/subcortical independence? We are certainly not dealing with a simple phenomenon which involves only isolated, specific regions of the brain. But it is a phenomenon well-suited to hypotheses and investigations aimed at understanding the link between brain activity, perception, cognition, and behavior.

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Pitch and Tone Quality

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Visiting Professor(Emeritus)

(1) The ***residue*** or ***residue pitch*** is the pitch of the fundamental that we hear when the fundamental is missing, or the fundamental and some of the lower harmonics are missing. This accounts for hearing the correct pitch over small transistor radios, or hearing the correct pitch near the lower end of the piano keyboard (the lowest note, an A, has a fundamental frequency of 27.5 Hz, which is just a low hum). Schouten first described the residue in 1940. Later workers found residues for just two sinusoidal components. But, if you play a scale with two sinusoidal components of constant frequency difference, you don't hear residue pitches; you hear a scale.

In 1972 Terhardt gave an elaborate formula for calculating what he called ***virtual pitch***, which is much like a residue pitch. I have found that his calculated virtual pitch isn't the pitch one hears for Risset's tone that falls in pitch when tape speed is doubled. Further, in 1960 and 1964 Flanagan and Guttman found that at pulse rates below 100 pps, pitches match for the same number of pulses per second, while for fundamentals above 200 Hz, pitches match for the same fundamental frequency. Yet, for a flat amplitude spectrum, but different phases, one can have either one pulse per period, or four pulses per period, three positive and one negative. Terhardt's virtual pitch would be the same for either phase spectrum; according to him, these patterns should never match on number of pps, but always on the fundamental frequency.

Something is amiss in the pitch literature, and I am trying to find what is up.

(2) Some years ago work of Max Mathews and me led to a new scale. Pieces in the new scale have been realized by Alyson Reeves, working with Max, and by me, through non-real-time computer synthesis. Jon Appleton has realized a piece on the Synclavier. I am having an overlay keyboard made for the Model II Synclavier, and perhaps for the DX7 keyboard. The Synclavier octave can be stretched to realize the new scale. The DX7 could be used to produce MIDI code to drive a synthesizer tuned to the scale. When I have the keyboard, I hope to persuade musicians to fool with it.

(3) Some time ago I found that "buzzy" or "electronic" quality of a synthesized tone can be avoided by deleting any harmonic components space closer than a quarter octave (an approximation to the critical bandwidth) apart. I hope to get back to this in a search for musically useful tones that have enough high frequency components to be bright, but yet are not buzzy or electronic in sound.

Analysis, Transformation, and Synthesis of Inharmonic Sounds for Computer Music Applications

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There are many ways in which a sound can be represented in a digital computer: time-domain representations (zero-crossing, autocorrelation), frequency domain representations (short-time Fourier analysis), homomorphic analysis (cepstrum), linear predictive coding, physical models, ...

The choice of the representation is dependent on the signal used and the application, therefore a careful study of all representations is necessary before making a choice. Here the concern is with computer music applications, that is, we are looking for sound representations valid for music composition with a digital computer. We want representations that will allow us perceptually accurate and very flexible ways of analyzing, processing, and synthesizing musical sounds, in particular inharmonic sounds.

Linear Predictive Coding (LPC) and the Short-Time Fourier Transform (STFT) are at present the most powerful and best understood techniques for the analysis/synthesis of signals. They are identity systems, that is, the signal can theoretically be recovered from the analysis parameters, and they offer some amount of flexibility for modifying the intermediate data before resynthesis. The STFT (also known as Phase Vocoder) has proved very efficient in dealing with harmonic sounds, and LPC techniques are used succesfully for speech-like sounds. Both techniques have been used for some time in computer music applications, but there is still very little analytical work asserting their potential or revealing their problems in this context.

This research focuses on comparing the two techniques mentioned, in the analysis, synthesis and processing of primarily inharmonic sounds. With the results obtained and considering the benefits and drawbacks of each one, some hybrid techniques are being studied to take advantage of both techniques. Parallel to the research studies is the design and development of a software environment that makes the work possible.

Linear Predictive Coding

In computer music LPC is considered a particular method under the more general category of analysis-based subtractive synthesis techniques (Moorer 1977). With LPC sound waveforms are modeled as an excitation function sent through a time-varying filter.

The usual application of this technique is with respect to speech comunication. The idea there is to reduce the data involved in the transmission of speech, and the goal is comprehensibility at the lowest cost. In musical applications the situation is very different. First of all, we would like to be able to apply such a technique to a more general group of sounds, not only speech. At the same time quality cannot be lost with respect to the original sound, and the objective is not to transmit the signal but to modify the sounds in ways that are useful for musical composition.

There is some work done on the use of LPC in Music (Lansky 1981, Moorer 1979). In particular, Moorer's article is a very good starting point for this work; it discusses some of the problems and finds solutions when LPC of speech is used in music applications. I am continuing his work, extending the family of sounds to which LPC can be applied.

Short-Time Fourier Transform

The STFT is an analysis-synthesis system that has as intermediate data the time-variant discrete Fourier spectrum of the input signal. Using the same terminology as Moorer (Moorer 1977), we can consider it a type of analysis-based additive synthesis. This technique has been more widely used in computer music applications than LPC and there are a few pertinent studies available (Smith and Serra 1987, Dolson 1983, Dolson 1986, Moorer 1978, ...).

Even though it was originally designed for speech sounds and with the same constraints as LPC—cheap transmission—it has already proved successful in computer music applications. It is very efficient for modeling and modifying stable harmonic sounds without fast transients, and there are some experimental extensions to the traditional algorithm in order to simulate inharmonic sounds. In particular this work uses the program PARSHL (Smith and Serra 1987) as the basic STFT representation from which further extensions are being developed.

Hybrid Techniques

Many inharmonic sounds can be understood as the sum of a series of non-harmonic partials plus a noisy component (Serra 1986). Given this model and considering the characteristics of LPC and the STFT, a compromise for simulating such signals is to model the deterministic components wih the STFT and the non-deterministic ones with LPC. Such a model requires a method of separating the deterministic from the noisy component. Because of these separate parts, some problems may arise in the process of resynthesizing such sounds. The development of this hybrid technique is underway by extending some previous work (Serra 1986), which was restricted to sounds produced by bar percussion instruments.

Tools for the Research

The research is being done on Lisp machine workstations (Symbolics LM-2) using the FPS-120B array-processor, and most of the basic programming tools have been taken from SPIRE (Speech and Phonetics Integrated Research Environment).

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Speech Research

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Speech sounds are important sounds. Since the dawn of man, humans have communicated (or at least attempted to communicate) with one another using vocal utterances. Vocal sounds are also important musical sounds. The singer is usually the focal point of a band, and many traditional instruments seem most "expressive" when they emulate vocal gestures in phrasing, pitch trajectory, or timbre.

Modern digital computers allow us to analyze and synthesize functions (in this case speech waveforms) with accuracy, repeatability, and computational speeds unavailable a generation ago.

Speech sounds have proved remarkably stubborn creatures, when subjected to intensive analysis and synthesis attempts. There are many reasons for this; perhaps a couple of them are:

1. Speech sounds change, and they change quickly. Additionally, their changes carry meaning (words, for example). Many analysis techniques assume a degree of stationarity present in the signal being analyzed which does not exist in the signal itself.

2. Speech sounds are very familiar. We understand our native language almost intuitively. We can identify many of our friends and "notable people" after hearing only a few seconds of speech.

Musicians have often found it convenient to classify tones with respect to their pitches, volumes, and timbres. With a simple tone, the notions of pitch and volume are fairly intuitive; one note is higher than another, or one note is louder than another. In each case the dimensionality of the parameter is assumed to be one (pitch is one dimensional) and time is considered to be irrelevant. Timbre is more elusive. Even for a static tone there is no "obvious" single timbral dimension. On closer inspection, the simple notion of pitch can get quite complex. A single note may be accompanied with glissando or vibrato (indeed, it is often these nuances that give a note "character"), turning a simple pitch number into a pitch trajectory that evolves in time. In a similar vein, the amplitude and timbre of many interesting sounds also evolve in time.

The purpose of this research is to attempt to quantify these trajectories for some useful signals, namely human speech signals. This involves primarily the number of available states for each parameter (for example, how many different volume levels) are needed, and how often they must be updated.

This is intimately related to "recognition" where it is necessary to map a very large class of input signals into a much smaller and finite number of output signals.

Particular attention is given to fidelity, and accuracy of parametric trajectories, rather than attempting to reduce bit rates while preserving intelligibility. For example, if the pitch trajectory is fast and complex, it is desired to capture its complexity rather than replace it with a more constant alternative which although still intelligible, may also sound very artificial.

Waveguide Digital Filters

Julius O. Smith III
Research Associate

Waveguide filters (WGF) have proven useful for building computational models of acoustic systems which are both physically meaningful and efficient for digital synthesis. The physical interpretation opens the way to capturing valued aspects of real instruments which have been difficult to obtain by more abstract synthesis techniques. Waveguide filters were derived for the purpose of building reverberators using lossless building blocks, but any linear acoustic system can be approximated using waveguide networks. For example, the bore of a wind instrument can be modeled very inexpensively as a digital waveguide. Similarly, a violin string can be modeled as a digital waveguide with a nonlinear coupling to the bow. When the basic model is physically meaningful, it is often obvious how to introduce nonlinearities correctly, thus leading to realistic behaviors far beyond the reach of purely analytical methods (Smith).

A basic feature of WGF building blocks is the exact physical interpretation of the contained digital signals as traveling ***pressure waves*** or ***velocity waves***. A byproduct of this formulation is the availability of ***signal power*** defined ***instantaneously*** with respect to both ***space*** and ***time***. This instantaneous handle on signal power yields a simple picture of the effects of round-off error on the growth or decay of the signal energy within the WGF system. Another nice property of waveguide filters is that they can be reduced in special cases to standard lattice/ladder digital filters which have been extensively developed in recent years. One immediate benefit of this connection is a body of techniques for realizing ***any*** digital filter transfer function as a WGF. Waveguide filters are also very closely related to "wave digital filters" (WDF) which have been developed primarily by Fettweis. Waveguide filters can be viewed as a generalized framework incorporating aspects of lattice and ladder digital filters, wave digital filters, one-dimensional waveguide acoustics, and classical network theory.

A ***waveguide*** for our purposes is any medium in which wave motion can be characterized by the one-dimensional ***wave equation***. In the lossless case, all solutions can be expressed in terms of left-going and right-going ***traveling waves*** in the medium. The traveling waves propagate unchanged as long as the ***characteristic impedance*** of the medium is constant. The characteristic impedance is the square root of the of the "massiness" times the "stiffness" of the medium; that is, it is the geometric mean of the two sources of resistance to motion: the inertial resistance of the medium due to its mass, and the spring-force on the displaced medium due to its elasticity.

When the characteristic impedance changes, ***signal scattering*** occurs, i.e., a traveling wave impinging on an impedance discontinuity will partially reflect and partially transmit at the junction in such a way that energy is conserved. Real-world examples of waveguides include the bore of a clarinet, the vocal tract in speech, microwave antennas, electric transmission lines, and optical fibers.

Waveguide filters are obtained (conceptually) by sampling the unidirectional traveling waves which occur in a system of ideal, lossless waveguides. Sampling is across time and space. Thus, variables in a WGF structure are equal ***exactly*** (at the sampling times and positions, to within numerical precision) to variables propagating in the physical medium in an interconnection of uniform transmission-lines.

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COMPOSITION

Introduction

Since the late 70's most of the work in composition at CCRMA has been done using a Foonly computer to control the powerful Systems Concepts Digital Synthesizer (built by Peter Samson and known as the Samson Box) for sound synthesis. The Samson Box is capable of utilizing many types of synthesis techniques such as Additive Synthesis, Frequency Modulation, Digital Filtering and some analysis-based synthesis methods. The software for controlling the synthesizer includes several languages (all based on SAIL) for patching together the synthesizer's 256 oscillators, 128 modifiers and delay memory. The very extensive Pla language (by Bill Schottstaedt) allows composers to specify parametric data for the Samson Box as well as for other sound processing procedures on the Foonly. The composer can use Pla simply by listing parameters and their values, or by creating algorithms which can range from the very simple to the unnecessarily complex for determining any number of parameters' values. Recently there has been a significant increase in the use of other computers and software for composition. Several composers have realized pieces which make extensive use of MIDI equipment, especially Yamaha synthesizers controlled via Macintosh computers. The acquisition of a Dyaxis Digital Audio Processor and several Macintosh II computers has brought renewed interest in software based synthesis and "musique concrète." The software being used for composition and developmental research for these systems include Le_Lisp, Smalltalk and C programming environments.

CCRMA has a long tradition of acommodating composers with diverse views about what music is and how to go about making it. A common link, though, is a dedication to an experimental field in which the results of one's work are often unpredictable. Many composers feel that this unpredictability can be of great advantage in the process of composing music and is, in fact, one of the main attractions of using computers for musical applications. Unforeseen effects of changing the value of a particular synthesis parameter are often incorporated into compositions and can even be the most interesting parts of a piece. Because the use of computers has shortened the time-span between composition and "playback," composing tends towards a more continual process of trial and error. Traditionally, composers of purely instrumental music wait for a performance of an already completed composition before making some of the kinds of decisions composers using computers often make about their work on a daily basis. As computers and digital synthesizers become faster and more powerful, the turnaround time between conception and realization of parts of pieces or entire compositions has become much shorter for some composers. The ability to work with real-time synthesizers (especially those with musical instrument interfaces such as keyboards) may bring with it, however, the temptation to leave many musical decisions up to the "inspiration" of the moment and to neglect some of the very important intellectual, non real-time aspects of music composition. Currently, "tape music" and music for live performer with a tape part are still the predominant types of compositions at CCRMA. But more composers are becoming interested in using real-time digital synthesizers, particularly because of the control they offer in a performance situation. Several composers are interested in the live interaction between live performers and synthesizers which is now possible with MIDI-based synthesizers and controllers. However, composers at CCRMA and elsewhere are still waiting for salvation in the form of an affordable real-time synthesizer that not only provides fast results and performance control, but also the generality, flexibility and programmability that larger non-real-time systems still offer—in short something like a real-time "software synthesis" system, with the possibility of applying many different types of synthesis techniques, compositional algorithms and performance directives. In the meantime, composers will, as always, make use of whatever sound-making devices (acoustic and electronic) are made available to them (or by them) to realize their compositions, and at CCRMA we are fortunate to have such resources as the Samson Box and the array of MIDI devices as well the systems now under development, on which to practice our art.

Since its beginning, works composed at CCRMA have been highlighted at music festivals, concerts and competitions around the world. Composers at CCRMA continue to be a major force in international New

Music events. Recently, compositions from CCRMA were performed at the annual International Computer Music Conferences in Vancouver in 1985, The Hague in 1986 and Urbana in 1987; at the Bourges Festival of Electroacoustic Music in France in 1985, 1986 and 1987; at the Gaudeamus Music Week in Amsterdam in 1987; at The Warsaw Autumn Festival in Poland in 1987; at the Cabrillo Festival in California in 1987; and in Argentina, Spain, West Germany, Sweden, Italy and Japan, as well as in many other concerts in the United States. CCRMA also produces its own annual concert series featuring new works from the center. These concerts are very well attended by the community, especially the yearly outdoor summer concert at Frost Amphitheater. Compositions from CCRMA have also won major electroacoutic music prizes over the past few years, including the Bourges Contest in France, the Luigi Russolo Contest in Italy, the NEWCOMP Contest in Massachusetts and the Irino Prize for Chamber Music in Japan. Recordings of works composed at CCRMA have been recorded on Compact Disks by Wergo and Harmonia Mundi; a second cassette of CCRMA pieces has been produced by the center; and Schott Music Publishers of West Germany are preparing to publish several pieces composed at CCRMA.

The following pages contain an overview of recent and current work in the area of composition at CCRMA and a list of recent compositions. A complete list of works created at CCRMA is available upon request.

contributed by Richard Karpen, ***DMA student, Music***

On "Leviathan"

Bill Schottstaedt

"Leviathan" is pure musique concrète. It does not expand the horizons of computer music; it wasn't even intended to! There are no illusory moving sounds, no timbre melodies, no MIDI commands, no new synthesis methods; nothing but simple, unassuming musique concrète. Were it not for the sheer tedium of making the 15220 splices required in the final version, there would have been no compelling reason to use a computer. Worst of all, "Leviathan" depends on reactionary tonal cliches embedded in unifying gestures that were exhausted before the beginning of the twentieth century. Is such backwardness permitted?

In the beginning I wanted to write orchestral music, music with lots of voices, large gestures, measured pace, simplicity; what used to be called the grand style. Such music has dubious precedents in this century, precedents laden with bombast and militarism. But as Longinus says [1], the fear of bombast leads to pedantry, and, if we leave aside gross incompetence, pedantry is certainly one of the distinguishing features of modern "serious" music.

Of course, no one writing orchestral music these days has any hope of hearing his music performed. Even if a competent performance is arranged, there is no reasonable way during rehearsal to fix unforeseen problems. The editing turn-around time averages four years or more; you get about ten tries to get it right, and then you die. The computer provides a way out—you get all the voices you want, and the turn-around time is usually measured in minutes or hours. Unfortunately, computer generated voices sound terrible.

Some composers say they are pleased with the racket they make. Why should one sound claim to be better than any other? Music is built from relations of sounds; the sounds themselves are arbitrary. The listener should put on his ears-of-the-future, ears that take pleasure in absolutely anything, as long as the program notes contain an incomprehensible treatise. But isn't the music of tomorrow simply tomorrow's music-of-today—what difference does it make what period of time the music belongs to? Will the "next aesthetic" really have so little relation to today's that we are debarred from judging music supposedly written for it? In any case, my ears are obdurate. They want to concentrate on the "aesthetic object," the thing intended by the sounds, the thing, whatever it is, that makes the piece hold together. From this point of view, the wretched sounds that computer musicians take for granted are no joking matter. Ugly sounds distract the listener from the music; they intrude and interrupt. The medium, the material basis of the work of art should not actively impede the presentation of that work. Our sounds should be good enough that we can ignore them. But try to synthesize a good sound. You'll quickly discover that synthesized sounds are ugly; they lack life; they cause fatigue; they suggest nothing.

What is it about a real sound that gives it its richness? We know that its steady state spectrum, assuming it has such a thing, is not very important. We know that its unsteady aspects are not simply a matter of filtered noise. We have been somewhat successful in mimicking long string sounds by using a broad, relatively flat spectrum, somewhat heavy reverberation, and slight (even imperceptible) vibrato. But even after years of experimentation, most real sounds remain a mystery.

"Leviathan" grew from one such sound, the groaning and creaking of a mast on a sailing ship. I could find no way to synthesize this sound without using far more words of data in the synthesis than were in the original recording. FFT's are useless here because each creak is so short (a tenth of a second or so) and the waveform is changing so rapidly that no window contains meaningful data. You cannot take the FFT of the sound, plug in an oscillator at each bin, and get anything like the original sound back. Phase vocoder (sliding FFT) methods fail miserably as well, even if you ignore the fact that they generate many times as much data as the original sound contained. FM techniques can come close, but they require too much of the synthesizer's resources (each "crack" of the mast takes about 40 oscillators). Rather than give up at the very beginning, I decided to use the recorded sound.

To get a broader palette of possibilities I also digitized a group of sounds similar to the ship sound: a dog howling, a rooster crowing, a horse neighing—all suggested by the groaning of the ship; a subway train, a diesel train with whistle, a steam engine starting up, a milking machine—suggested by the creaking sound and the wind; and finally Bob Shannon and myself making "Godzilla"screams. These sounds turned out to be so rich in possibilities that I never felt any need for live or synthesized voices. In fact, "Leviathan" barely scratched the surface of these sounds. Every day I found myself confronted with entirely unexpected, beautiful sounds. The whales, for example, fell into my lap when I mixed a bunch of dog howls with a breathing sound from the screams; the choral effects came from a portion of the rooster crowing; the strings came from the mast creaking. There seemed to be no end to the riches buried in these recordings.

To get the steady pitches, a somewhat stable portion of the original file, say a tenth of a second, was enveloped to avoid clicks, then overlapped with itself at irregular intervals (regular repetition produced an irritating tremolo). The more overlapped these repetitions, the more "reverberated"the result. Once one steady pitch was available, sampling rate changes (using linear interpolation) provided arbitrary pitches. These slowly organized themelves into a basically diatonic, tonal fabric warped by unusual tunings in a manner nearly identical to that used in my last several synthesized pieces (in particular, the first movement of "Water Music").

The use of extended tonality was deliberate, as was the reliance on string-like and voice-like sounds. Of course, the Learned Listener's response is "This is not new music—this is just a neo-romantic pastiche," and he stops listening. Gadamer remarks, "Ease of assimilation obstructs the hermeneutical process"[2]. That is, an unproblematical surface makes a certain class of listeners tune out. But the surface this group demands (percussion-based serialism or chance music) is opaque to the vast majority of the audience. The Learned Listener, of course, would claim that this opacity is rooted in the complacency and dull wits of listeners other than himself, but it could also be the case that the twentieth century's notion of "serious" music is a morass of premature systemizing, bad logic, and simple foolishness. If you believe the theorists, a "logical" score (read numerological, or even paralogical) is necessary and sufficient to guarantee that the music is good. But despite all the windy theorizing and pounding on the table, despite all the clever musical "just-so stories" we get fed in history books, despite all the social pressure imposed by true believers in whatever is the latest rage, most of these pieces, whether based on some arbitrary "method," or on borrowings from pop-mythology (eastern mysticism, for example), simply cannot stand on their own as music. The unity, the logic of music is not so simple that a pretty graph or a simple schema can embody it. But, leaving aside problems with the current notions of "serious" composers, I see the decision to write in extended tonality as primarily a political choice—the political (or social) function of art is to bind together the social group, just as its moral (or religious?) function is to bind the individual to the world. From this point of view, in normal circumstances the artist should speak the vernacular. It bears mention that today's musical vernacular is a far richer language than the language of the academic elite, so this decision is actually one toward increased complexity and greater demands on the composer.

The choice of diatonicism, on the other hand, is more a matter of personality. I just feel more comfortable with it; at some point, you have to trust your taste. My dissatisfaction with pitch (or interval) comes from tuning problems outside equal tempered twelve tone tuning. A good example is "Dinosaur Music." Here nearly every dissonant chord is based on quarter tones—by squeezing minor seconds and stretching fourths I was able to get some intriguing and intense dissonances. Nearly every piece since "SandCastle" has explored non-standard intervals in search of dissonance, consonance (Pythagorean and just intonation in "Water Music"), and intense voice leading (the most obvious example being the raised leading tones in "from the Book of the Burning Mirror"). Without a computer such exploration is tedious, if not practically impossible. The composer either has to build his own instruments (Harry Partch, Johannes Goebel), or try to find players who can learn the new pitches and notations. Both of these approaches involve long turn-around times. As with orchestral music, the edit cycle (or experiment cycle, if you like) is simply too long. On the computer, however, the composer can try new pitches quickly and easily. He is not tied to any one system either by instrument design or performer training. Of course, these new dissonances demand resolution; the entire fabric of our music warps itself to these demands; we end up using "out of tune"pitches all the time. My goal in all this is to try to hear what the music itself intends, not to provide a little cubby-hole for every pitch (as in the musical cross-word puzzles of Babbitt and friends). Our musical decisions should bring the "intended object"as clearly as possible into the presence of the listener.

The first such decision in "Leviathan"was that it would be monaural. Since the days of the first "high fidelity" recordings, stereo has been the norm. The argument is that stereo makes it easier to separate the musical voices, and quad provides a more convincing illusion of movement. In my opinion, a moving sound is simply a distraction, a gimmick, and spatial placement doesn't matter. The performers should sit together and sit still. I have never seen, or heard of any case in which placement or movement was crucial. In pieces that make a big point of placement, the separation is simply an irritation—you get the feeling that some of the players missed the bus and are out in the lobby trying to play along. And despite the optimism or innocence of some composers, a factor like sound placement cannot be made to matter in music simply by fiat.

Are performers even needed? Why do concert-goers find a pure tape piece unsatisfying? We can back up a little, and ask: why have concerts at all? We take it for granted that we are not interested in the athletic displays of virtuosos. One plausible excuse for a concert is that it provides an organized occasion for the listeners to listen intently and without interruption [4]. Since one aspect of music is play [5], the listener takes advantage of the quiet intentness of the concert setting to play along with the players, to participate in the play of the music by watching the musicians. In tape music, there are no players—the listener must participate by listening, by following the play of thought embodied in the sounds, something that is apparently much more difficult. Perhaps some of the difficulty is based on the notion of expectations—the composer assumes his listeners have a certain background, a shared knowledge of past music for example, and much of what he is doing plays with those expectations. But in the modern concert world, listeners have very different backgrounds, and to some extent different expectations, so some of the playfulness may seem willful, arbitrary, or senseless. The listener "gets lost"(this by the way provides an explanation for the proverb "de gustibus non est disputandum"[7], and provides a plausible setting for music theory as an act of cultural education [8]). It is not a matter of indifference to me that some listeners get lost—I have never understood those composers who claim they feel contempt for their audience. In the contemporary setting audiences are almost too open-minded, not to mention polite (as opposed to the 1920's in Vienna, for example), so this contempt looks like self-hatred turned outward; from another point of view these composers are merely showing their own lack of craft. But then, as Dickens says, "If they are bad, think that they would have been better, if they had had kind friends, and good homes, and had been taught better"[9].

After deciding that "Leviathan" would be monaural, the next decision was to use a predominately string-based texture. We want our medium to be transparent—it should not call attention to itself. But in the upside down world of new music, even the use of a transparent medium is a stumbling block. "Muzak," mutters the Learned Listener, and he shifts uneasily in his seat. But if he will stay and listen (something we have good reason to hope for), I think that the music will speak even to him, and by speaking the vernacular we also retain the less patient portion of our audience (the portion that might otherwise get lost). Then, if we have succeeded as composers, the music "in" the music will be present for everyone. But to have even the hope that this can happen, we have to strive for transparency. I apply this criterion to nearly every aspect of the material presentation of music. For example, in a good performance the performers disappear. The romantic, intense, virtuosic phrasing that is so popular these days is simply the performer saying "look at me," when he should be saying "listen to this." Bad phrasing is exactly like a bad sound, or an unconvincing transition—it is a form of musical falsity. Truth being correspondence or adequacy of the presentation to the thing coming to presence [11], of the concept to the object of the concept [12], anything that obstructs that presence falls short of truth.

Some listeners apparently do not hear anything "in" the music—they claim that music is just a pleasing wash of patterns of sounds. They connect a voltmeter to an amplifier and say, "Now where is God in my voltmeter?." Of course I can't prove the "musical thought"is there to someone who has decided to play the skeptic. In fact I can't even describe it in words in such a way that other equally knowlegeable listeners would agree with me, but why does even our skeptic find one piece more pleasing than another? He cannot fall back on saying that good music is a clever handling of the sound patterns, because the "cleverness"is exactly what is in question here—the new word is merely a verbal ploy, a begging of the question. He could not even talk about patterns were he not idealizing each sound and each pattern, for how could one pattern match another were not both idealized or abstracted to remove the "unimportant" differences? Surely every sound differs from every other, especially in normal performance situations, so the listener must be remembering, or at least matching patterns with an idealized version of what he has heard. As Kierkegaard points out [13], there could not even be repetition without idealization. The direction of that idealization is based on expectations formed historically. So, already our psychologist is carrying on an active form of interpretation of what impinges on his ears. General Grant didn't realize what a complicated thing he was doing when he recognized "Yankee Doodle"[14]. Once the General started idealizing (remembering what he heard and making comparisons), what was he doing?

Beethoven called it "thinking in tones"[15]. Now when Beethoven held a piece in his mind, can our psychologist claim that there was nothing there? He would say music exists only in the sounds, but clearly music exists in whatever form the composer thinks it in. The sounds merely embody that thought. To communicate it to another person, to embody it in a material form that is independent of the composer and that might outlive him (unlike the material form stored in his memory), the composer writes it down. Once embodied in sound, music is still more than that sequence of sounds, more than a layering of such sequences—good music somehow makes these layers cohere. The "real" music is in the focus, energy, clarity, concentration of thought, or whatever makes it one thing, not in the structure sketched out by patterns. "My God! What has sound got to do with music! That music must be heard is not essential—what it sounds like may not be what it is."[17] It is music that makes sense of sounds, not vice versa [16]. And the fact that some sense experience (sound) is needed to know that something else (music) exists is not an indication that that thing (music) is purely, or even primarily a matter of that sense experience.

Several philosophers (Schopenhaur for example [18]) have emphasized the immediacy of music, and I certainly am not trying to downgrade the impact of the sheer beauty of sound. But that impact is worn away by time—what one time hears as immediate beauty (to some extent this can be gathered under the title "popular music"), another period does not notice at all; the music that depended purely on this impact comes to sound naive or comical. It is sometimes difficult to pull back from the popular music of one's own time simply because of this immediacy, but as a substitute one can immerse oneself in the popular music of an earlier time. The salon music of the late nineteenth century, for example, still exists in scores buried in basements and archives, and with a little effort you can begin to hear what they were hearing. But the only music that survives from that time and still speaks to us has something more durable. What is it that makes one piece live while others die?

Plato said that art makes the idea visible [19]; Kant preferred the word "form"[20], saying that art is intent on form and the sensuous stimulus merely carries that form. Hume mentions utility [21], the beautiful is that which is suited to its purpose of giving pleasure, but as Kant points out [22], pleasure is not the main effect of beauty, much less its purpose. Kant goes to some trouble to show that our response to beauty is "disinterested"[23]; we recognize that the surface, the sensuous stimulus, is not what is important. Gadamer tries to bring this out by saying that a performance "only succeeds in communicating a genuine artistic experience of the work itself if with our inner ear we hear something quite different from what actually takes place in front of us" [24]. This thing we hear with our inner ear is not unique to each listener. We might use the phrase "aesthetic object"[25] to emphasize that there is something there that can be heard by anyone who listens. Certainly any work of art leaves a lot up to the listener. Not only must he recreate the aesthetic object, but in every work of art there are "regions of indeterminacy"[26] where one's imagination is free to fill in. It may be, for example, that real sounds are richer and more alive than synthetic sounds because they provide more room for the listener's imagination. But in a successful work of art, the aesthetic object shines through any such filling-in; the aesthetic object speaks across vast distances of time and culture. What I have been calling "transparency" is an embodiment that brings that object to presence despite the semi-blocked ears of the listener. And by speaking the vernacular (the current common music practice), one sidesteps blockages based purely on unfamiliarity with the language, on the composer's part as well as the listener's.

The Learned Listener now grumbles that new music should try to stretch the bounds of current practice, but what is the basis of this demand? I have yet to see any explanation that is not simply silly, and isn't it suspicious that the music of the "avant garde" is nearly identical to so-called "pop music"? My hunch is that musical practice in general is a somewhat random rambling about in the space of possible musics; each period choosing some set of conventions for what it thinks is a pleasing sound. The artist accepts that conventional background and works within it. There is no particular point in writing in the style of some other time; there is even less point in inventing new conventions according to some intellectual notion of what music might be [27]. Music itself might somehow drive one toward new conventions, but this is not a process that can be predicted or channeled into systems. We could almost say that music is "there" independent of the composer or his listeners, and the composer merely discovers some aspect of it from the currently fashionable viewpoint. The work of art then brings that aspect to visibility and gives it a place to dwell (to borrow from Heidegger [28]). Writing music has nothing to do with self-expression; it is not simply a matter of inventing structures; it is not a form of self-therapy; it is not a job; it is only indirectly a platform for a moral or social forum; and it is an act of science only in the sense that both science and art are explorations of reality (that which is independent of what I wish it were).

"That Crawford Tillinghast should ever have studied science and philosophy was a mistake" [29], the gentle reader must be muttering, so I will return to prose. Thinking in tones, like thinking of any kind, is subject to any number of blockages. Clarity and focus are not automatic. I often feel that my ears are clogged by previous pieces; old habits, personal cliches constantly beckon, saying "here's a tried and true way out"—when in doubt, hit a drum or hold a note; it becomes harder and harder to "say the same things about the same things" [30] without simply repeating myself like a fool. To overcome such blocks, many composers improvise. The trap here is that your muscles start to do the thinking—you play what falls under your fingers. The computer provides an interesting alternative—the composer can set up compositional algorithms, that is little programs that generate music, and then listen to the result. There is often something that can be saved for future use. One of the first things I came upon in this regard was that tightly bound gestures can grow from mensuration canons in many voices, that is, canons in which each voice has its own tempo. The first movement of "Colony," for example, is made up almost entirely of twelve-voice canons of this kind. It would be nearly impossible to notate or perform these canons because the tempos are not related in any obvious way (one voice's quarter note is another's 1.03 quarter notes for example), but it is easy to hear what is going on.

The notation of the compositional algorithms is a computer program. This in itself is a welcome change from common music notation and the varieties of proportional notation. A great deal of modern music is eye music—the composer is thinking notation, not music, even to the pathetic extent of thinking that his new notation will lead to new music. Our desire is simply for an open-ended, neutral notation; the less it suggests, the better. Standard programming languages, which have grown from the vast experience of programmers, are sufficient for all a composer's needs—there is no need for a special language such as Pla [31]. One could even argue that such a language is an intrusion because it makes certain kinds of musical thinking easier to express. There is nothing in music that is hard to program except perhaps scheduling in real time. Since "Leviathan" had no expectation of running in real-time, the programming involved was trivial. From the composer's point of view, computer science issues such as parallelism, language design, real-time control, user interfaces, and so on are uninteresting and unimportant. Think music is his imperative. Anything that actively impairs that thought is bad. The composer's computing environment should make it possible to ask arbitrary questions, and the faster he can get answers, the better. But, leaving aside the happy accidents encountered during improvisation, what is the source of the questions?

My starting point in many pieces is the feeling that justice has not been done to a sound, that the sound is begging for a setting. In "Leviathan" that sound was the creaking of a ship. Because it was all but impossible to deal with this sound on the synthesizer, musique concrète was unavoidable. The musical language chosen was simply the vernacular at the time. The medium was completely decided by considerations of transparency.

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7. *"There's no disputing about taste".*

8. E.g. Jos Kunst. "Remarks on Analysis," *Interface* 1987, 16:1-11.

9. Dickens, Charles. "Life of Our Lord," Chapter 3. (Quoted in the introduction to the first volume of *The Christmas Books,* Penguin Books, 1982).

11. from Heidegger. See for example, "The Origin of the Work of Art," in *Poetry, Language, Thought,* New York: Harper and Row, 1975, Albert Hofstadter, trans.

12. E.g. Popper, K R P, *Conjectures and Refutations,* "Truth, Rationality, and the Growth of Knowledge," New York: Harper and Row, 1965, sect 2, p223). Also Aristotle, *Metaphysics* B, 2 996b26-30.

13. Kierkegaard, S. *Johannes Climacus,* "What Is It to Doubt?," Princeton, NJ: Princeton University Press, 1985. Hong and Hong, trans. p171.

14. This is a joke I've heard many times, but can't trace (Grant told Lincoln he could recognize two tunes; one was Yankee Doodle and the other wasn't).

15. I cannot track down this quote – maybe it's from Leonard Ratner.

16. E.g. Wittgenstein, *Remarks on Color,* Berkeley, CA: University of California Press, 1977. sect 213 and 293.

17. Ives, Charles. *Essays Before a Sonata, The Majority, And Other Writings,* New York: W W Norton, 1962, p84.

18. E.g. Schopenhaur, A. *The World As Will And Representation,* New York: Dover. Vol I, part 52, p255, or Vol 2, chap 39, p447. Also a brief note in *Parerga and Paralipomena,* Oxford, Clarendon Press, 1974. E F J Payne, trans. Vol I, "Fragments for a History of Philosophy," p37, and Vol 2, chap xix "On the Metaphysics of Music," p415.

19. Plato. *Phaedrus,* 250d. (Aristotle agrees by saying that poetry is more philosophical than history, *Poetics,* 1451b5).

20. Kant, I. *The Critique of Judgement,* Part 14, p65 of the "Analytic of the Beautiful," for example.

21. Hume, David. *A Treatise of Human Nature,* New York: Penguin Books, 1987. Book III, Sect 1, p627.

22. E.g. Kant, *The Critique of Judgement,* Book I, sect 13, p64.

23. Kant, I. *The Critique of Judgement,* sect 5, p44-45.

24. Gadamer, H. *The Relevance of the Beautiful,* p44.

25. Borrowed from Ingarden, R. *The Cognition of the Literary Work of Art,* Evanston, Ill: Northwestern University Press, 1973. R Crowley and K Olson, trans. See p83, 85.

26. Ingarden, Roman. *The Cognition of the Literary Work of Art,* part 11, p50, *Concretization of the Portrayed Objectivities.*

27. E.g. Roads, Curt. "Interview with Herbert Bruhn," in *Composers and the Computer,* Los Altos, CA: William Kaufmann, 1985.

28. E.g. for example, Heidegger, M. "Building Dwelling Thinking," in *Poetry, Language, Thought.*

29. Lovecraft, H P. "From Beyond," in *The Lurking Fear and Other Stories,* New York: Ballantine Books, 1971.

30. A loose paraphrase of Socrates, quoted in Plato, *Gorgias,* 490e and Xenophon, *Memorabilia,* iv, 4, 6 (p311 of Loeb edition): *"always the same, and – what is more astonishing – on the same topics too!"*

31. Schottstaedt W G. "Pla: A Composer's Idea of a Language," *Computer Music Journal,* 1983, 1:11-20.

Schottstaedt W G. *Pla Reference Manual,* Stanford, STAN-M-19, 1984.

Works in Progress

- A collaboration with composer Dexter Morrill and jazz musician Wynton Marsalis for a piece for trumpet, bass, keyboard and computer which will be filmed for public television.

- A work for computer and digital keyboards by composer John Chowning.

- A commission for a choral work with tape by composer David Jaffe.

- A work for oboe and tape by composer Amnon Wolman for Heinz Holliger.

- A work for orchestra, MIDI keyboard and tape by composer Stanislaw Krupowicz.

- A work for string quartet and tape by composer Heinrich Taube.

- A new piece by Richard Karpen for four performers of Yamaha synthesizers (A MIDI Grand Piano, 3 DX7II's and 2 TX802's) at the request of the Juilliard School. Performance scheduled for May 6, 1988 at Lincoln Center in New York.

- A commission from the Tibia Quartet of Amsterdam for a piece for four flutes and tape, by composer Richard Karpen.

- An opera theater piece by composer Amnon Wolman.

- A musical theater piece by composer Janis Mattox.

Works Completed at CCRMA from 1985 to 1987

BERGER, Jonathan - Island of Tears, quad tape, 9 min. 1985.
BERGER, Jonathan - Meteora, 9 min., 1987. 2nd Prize, Concours de Bourges, 1987.
BOUGHTON, Rachel – Pigeon Fantasies, movie sound track, 1985.
CHAFE, Chris - Quadro, for piano trio and tape, 17', 1986.
CHAFE, Chris - Virga, for harp and electronic harp synthesis, 1987.
DAL FARRA, Richardo - Untitled, computer-generated quad tape, 1986.
DELACRUZ, Zulema - So Far, So Near, computer generated tape, 7 min., 1987.
FIELDS, Matthew - Walk, for quad tape, 20 min., 1986.
FULTON, Douglas - Tip the Velvet, tape, 7 min, 1987.
FULTON, Douglas - Bowling for Blood, tape, 6 min, 1985.
FULTON, Douglas - You'll Never Walk Again, tape, 6 min, 1985.
GOEBEL, Johannes - Passage (- nur ein einziger Akt -) , for microtonal mallets, speakers and computer-generated tape, 50 min., 1982-1986.
HOLLAND, Anthony - Suite, computer-generated quad tape, 1986.
HALLSTROM, Jonathan - Telos, for solo piano and computer-generated sounds, 1987.
JAFFE, David - Grass, for female chorus and stereo tape, 1987. 8 min. Commissioned by The Skidmore College Choir.
JAFFE, David - The Fishing Trip, for male chorus and stereo tape, 1986. 7 min. Commissioned by Chanticleer.
JAFFE, David - Telegram to the President, for string quartet and stereo tape. 5 min., 1985. Commissioned by the Kronos Quartet.
JAFFE, David - Impossible Animals, for chorus and quad tape. 10 min., 1986. Commissioned by the Hamilton College Choir.
KARPEN, Richard - Eclipse, for computer-generated tape (two or four channels), 20 min, 1986. Awarded First Prize at the NEWCOMP International Competition, 1986; Honorable Mention at the Bourges Electroacoustic Music Contest 1987.
KARPEN, Richard - Exchange, for solo flute and tape. 13:10 min 1987. Awarded Bregman digital music

award at Bourges Electroacoustic Contest, 1987. Honorable Mentions at 1987 NEWCOMP and 1987 Luigi Russolo.

KARPEN, Richard - Il Nome, for soprano and tape, 1987. 14:40 min. Commissioned by Judith Bettina.

KARPEN, Richard - Idioma, for tape, 1987, (solo tape version of Il Nome). 14:40 min.

KRUPOWICZ, Stanislaw - Thus Spake Bosch, for quad tape, 14 min., 1985. Second Prize at the NEWCOMP International Competition, 1985.

KRUPOWICZ, Stanislaw - Farewell Variations on a Thema by Mozart for amplified string quartet and tape, 26 min., 1986.

KRUPOWICZ, Stanislaw - Farewell Variations on a Thema by Mozart, version for computer-generated tape, 26 min., 1986.

KRUPOWICZ, Stanislaw - Half Way Through, version for tape, 26 min., 1986.

KRUPOWICZ, Stanislaw - Concerto for Sax and Computer, 1987.

KRUPOWICZ, Stanislaw, MILOSZ, Tony, MOWITZ, Ira - Nightfall, 40 min., 1987.

MALOUF, Fred - Chromatonal, for quad tape, 13 min., 1985.

MALOUF, Fred - Rama's Return/Sacrifice of the Horse, 7'40, 1986.

MARIN, Servio - Retour au Silence, for quad tape, 9 min., 1985.

MARIN, Servio - Visonual Reverie, for soprano, trombone, piano, and tape, 10 min., 1985.

MARIN, Servio - Fantasmas de dos mundos, for dance, actors, computer music, and sculpture, 1987.

MCNABB, Michael – Invisible Cities. ballet music for piano, saxophone, computer tape, and electronics, 40'., 1985.

MOWITZ, Ira - Jubilum, stereo tape, 20', 1986.

MOWITZ, Ira - Darkening, stereo tape, 11', 1987.

NUÑEZ, Adolfo - Canales, quad tape, 10' 54", 1985.

NUÑEZ, Adolfo - Press , quad tape, 9' 5", 1986. 2nd version.

NUÑEZ, Adolfo - Images. For mezzo-soprano, piano, French horn, trumpet, trombone and stereo tape, 13 min., 1986.

POOR, Robert – Secret and Profound, quad tape, 1985.

ROOK, Victor - Itching, quad tape, 1986.

SCHOBER, Brian - Voices, quad tape, 16 min., 1985.

SCHOTTSTAEDT, Bill - Water Music, quad tape, 10 min., 1985.

SCHOTTSTAEDT, Bill - Sonata, quad tape, 7 min., 1986.

SCHOTTSTAEDT, Bill - Put on a Happy Face, quad tape, 7 min., 1986.

SCHOTTSTAEDT, Bill - Leviathan, computer-generated monophonic tape, 7 min., 1987.

SCHOTTSTAEDT, Bill - Brand X Music, 9.5 min., 1987.

SCHOTTSTAEDT, BERGER, JAFFE, FULTON, SHANNON - Fanfare, 5 min, 1985.

TAUBE, Heinrich - Jubjub, quad tape, 11 min, 1985.

TAUBE, Heinrich - Tremens, quad tape, 1987.

USSACHEVSKY, Vladimir - computer-generated sound for analog tape piece, 1985.

WOLMAN, Amnon - Perhaps, at last, Some such hours passed, quad tape, 8.40 min, 1985.

WOLMAN, Amnon - Mora, for soprano, mezzo-soprano, oboe, percussion and stereo tape, 16 min,1985.

WOLMAN, Amnon - If Thorns..., computer-generated stereo tape, 12 min, 1985.

WOLMAN, Amnon - A Circle in the Fire, for bass clarinet and computer-generated stereo tape, 13 min, 1986. Dedicated to Harry Sparnaay.

WOLMAN, Amnon - M, for orchestra and computer-generated stereo tape, 16 min, 1986.

WOLMAN, Amnon - And then she said for actress, four pre-recorded voices, computer- generated sounds, and computer graphics. 25 min, 1987.

WOLMAN, Amnon - Ladders and Plains, computer-generated stereo tape, 25 min, 1988.

Program Notes

"Island of Tears" (JONATHAN BERGER)

Recent performances:
Humanities Conference, Stanford University, April 1987
International Music Festival, Bourges France 1986.

Immigrants awaiting permission to enter the United States branded Ellis Island, the holding area for aliens in the early years of this century, 'An Island of Tears'. The work is a tribute to my own immigrant roots and to the plight of today's refugees and exiles.

"Meteora" (JONATHAN BERGER)

Winner of second prize for digital music at the 1987 Bourges Electroacoustics Music Festival in France.
Recorded for Harmonia Mundi, Compact Disc, 1987 with Groupe de Musique Experimentale de Bourges.
Scheduled for publication by Schott of West Germany in 1988.
Recent performances:
SEAMUS Conference, Dartmouth College, New Hampshire, October, 1987.
International Music Festival, Bourges, France, June, 1987.
Computer Music Festival, Stanford, July, 1987.

"Meteora" is a monastery near Delphi built on the summits of three enormous erratics dramatically set on an otherwise flat landscape.

"Virga " (CHRIS CHAFE)

For Harp and Electronic Harp Synthesis.
Performances:
Stanford, 1987
Warsaw Autumn Festival, Warsaw, Poland, September 1987.
Festival of Soloists, Macerata, Italy, July 1987.

The title comes from the name of a cloud formation that I've seen out in wide-open spaces while travelling across the western U.S. Curtains of rain hang halfway to the ground from wispy clouds, an appearance that is caused by the moisture evaporating on the way down.

The duo is a solo for two players. Their combined sounds create an instrument that is larger physically and timbrally than the acoustic instrument alone. The drum gives cues to a synthesizer programmed to generate harp sounds. Its score and cue patterns are prestored in the synthesizer. The drum and software for the system were developed by Max Mathews.

"Improvisation" (CHRIS CHAFE and DEXTER MORRILL)

For Trumpet, Celleto and Electronics.
Performed at CCRMA (informally) and Feb-88 festival at Clark University.

Musicians have the possibility, more and more, to work with interactive music synthesizers. Fairly powerful computers can be brought on stage which don't require air conditioning or soundproofing (at least for the moment). The improvisation uses synthesizers have been programmed to accompany and react, using material played by the musicians. The music has a number of sources: the Yamaha TX-802, stored instrument tones from the Prophet 3000 sampling device, and the two instruments themselves. The piece consists of some precomposed or preperformed music, some textures that are triggered by the computer and which serve as material for the two musicians to perform with, and an interactive computer MIDI program which Morrill calls the interactive duet loops. Equipment is all off-the-shelf. Programs by Perry Cook and both the players are used.

"Quadro" (CHRIS CHAFE)

Previous version performed Stanford, December 1986, San Francisco 1987.
(this version to be recorded by Jefferson Quartet, April 88)

The material of Quadro grew out of experiments in fusing recorded sounds with artificial sounds. Using digital audio editing programs, short sound samples were spliced onto computer generated tones. A variety of recorded instrumental articulations, as well as alterations of splice timing and abruptness, created some interesting expressive controls. A computer process was used to string the "hybridized" notes together into phrases. These mixtures were swirled around in pitch and time (marble fudge style) changing in density and rhythm over long phrases. The computer sounds accompanying the trio are a result of exploring different recordings, mainly a cello part from an earlier piece and some loose licks caught during the tune up before an orchestra concert. The instrumental score was added later. The string and piano parts were written with the same interest in swirling together distinct timbres and lines.

"Walk" (MATHEWS FIELDS)

Performed at Stanford, July 1986.

The traveler wanders through innumerable scenes and suprises, encountering strange and wonderful people along the way. S/he is accosted from time to time by wild cellos, sometimes singly, and sometimes roaming in packs; various other surreal phenomenon occur with or without warning. A babbling brook brings to mind a haiku by Matsuo Basho:

an old pond;
as a frog jumps into
the sound of water...

It is never at all clear whether the path leads through the great mysteries of nature or through the more personal mysteries of the mind.

WALK incorporates a variety of software developed by the enterprising students and associates at CCRMA. Among the musicians and programmers whose contributions helped are Jan Mattox, David Jaffe, Bill Schottstaedt, Paul Weineke, John Chowning, Leland Smith, Douglas Keislar, and Tovar.

"Passage (- nur ein einziger Akt -)" (JOHANNES GOEBEL)

With words, loudspeakers and different instruments for percussionists, speakers and electronics.
Performances:
Hannover, West Germany, 1985, 1986.
UCSD/CME, 1986.
Stanford, July, 1986.
Kassel, West Germany, 1987.

The German title of the piece "nur ein einziger Akt" is a quote from the Kafka text recited during the performance, the translation being "just a single file." The German title has a variety of meanings if it is taken out of its context. Possible meanings of "Akt" are: an act in a theater-play, a painting of nudes, a file (but only in the concrete meaning as an assemblage of papers, "Akt" is not the equivalent to "file" in the computer world), sexual intercourse, a proceeding limited in scope.

The sounds produced via loudspeakers were composed at CCRMA during the summer of 1983. The voices of the speakers are reproduced via loudspeakers. The percussionists play live with and on instruments built by the composer.

The percussionists play in an unusual world: on their mallets the highest keys are in the middle, to the left and to the right descending scales.

The tuning system of the instruments differs from "ethnic" and "western" systems in having no cyclic division into octaves. Pitches do not form pitch-classes but the whole frequency range is divided by interval-classes which do not result in common pitches of the different scales. Each interval is the sum of the previous two intervals.This, of course, is the Fibonacci-series, the approximation of the Golden Mean. For sure, one cannot hear the proportions of the Golden Mean as one can see them (as it is equally impossible to hear symmetry in the layout of a sonata). But a unique set of intervals creates an unmistakable environment. Especially, the intervals of each scale played in sequence are identifiable, as the very narrow steps widen into bigger and bigger leaps.

In his piece "Stria," John Chowning was the first to use the Golden Mean in a musical way at the macro- and micro-compositonal level. The application of the Golden Mean in music has a tradition, but John Chowning's synthesis method of frequency-modulation yielded itself readily to this compositional approach; the scale was in part determined by the Golden Mean, set in relation to the spectral components of each pitch by utilizing that property of FM which distinguishes it from other synthesis methods. The construction of "real" physical musical instruments with a tuning system governed by the Golden Mean was inspired by John Chowning's work.

Naturally, the timbre of the mallets still obey the physical laws that apply to bars. So the consequence had to be to go back to the point of departure, back to the computer. Having constructed a scale which is different from Chowning's (in this piece, the ratios of the intervals being determined by the Golden Mean) additive synthesis was used to create the timbre whose frequency-components reflected the intervals of the scale. Again a speculative approach (or pragmatic—both ends meeting in infinity, which is close to home): the horizontal microstructure of timbre corresponding to the vertical macrostructure of scales, simultaneous and sequential events being crosswise linked together.

In the digital medium it is no problem to transpose the scales, to create a new scale based on any pitch of the original scale embodied in the mallets. Still, this would have been possible in the physical world as well by building scores of xylophones. But the interpolation between timbral structure and tone shape is only possible in the digital domain. (The shape always reminds one of some known acoustical event—"That sounds like an organ, that like a voice," etc.—whereas the timbre determines the flow of association usually only in correspondance with the shape—e.g. "It sounds like a funny violin.")

The rhythmic values are mostly also proportioned by the Fibonacci series. This poses certain problems to the percussionists; it requires a new mode of thinking and counting, playing and feeling. The computer does not care.

"Telos" (JONATHAN HALLSTROM)

For piano and tape.
Premiered at SEAMUS Conference, Dartmouth College, New Hampshire, October 24, 1987

"Telos" was composed while I was on a Rockefeller residency at CCRMA during the summer of 1987. All the sounds were derived using FM techniques, implemented with William Schottstaedt's composition language PLA. The basic intent of the piece is to provide a digital counterpart to the live piano; close, but not exactly matched to the sound of that instrument, which is then acoustically manipulated in a dialogue-like exchange where the computer sounds not only interact with the piano as a second performer, but also provide amplification of its various resonance characteristics.

"Grass" (DAVID JAFFE)

For female chrous (SSA) and computer-generated and processed tape - 7.5'. Commissioned by the Skidmore College Chorus.
Performance: Dec. 1987, Saratoga Springs, N.Y.

"Grass" (1987) is the third of a series of pieces for chorus and tape. Unlike its predecessors "Impossible Animals" and "The Fishing Trip," "Grass" is not based on an original text. Instead, the text is the Carl Sandburg poem by that name. This marks the second time Jaffe has set Sandburg's poetry. The first was in "May All Your Children Be Acrobats" (1981) for eight guitars, voice and tape, based on excerpts from "The People, Yes."

"Grass" is concerned with the process of recovery, in particular with the healing of the wounds of war. It was commissioned through a gift from Lincoln and Gloria Ladd for the Skidmore College Chorus.

"The Fishing Trip" (DAVID JAFFE)

For 12-voice male chorus and computer-generated and processed tape - 8'. Commissioned by Chanticleer.
Performances:
Nov. 1986, Sacramento, CA. (part of American Music Week).
Jan., 1986, Herbst Hall, San Francisco, CA.
Palo Alto, CA.
Concord, CA.

"Impossible Animals" (DAVID JAFFE)

For chorus (SATB) and computer-generated quadraphonic tape - 8' Commissioned by Hamilton College Choir.
Performances:
June, 1986, Stanford, CA. (preview)
Nov. 1986, Hamilton College, Clinton, N.Y. (part of American Music Week).

"Impossible Animals" is an antiphonal interplay between human and synthetic singers of various sexes and species. The text is transcribed verbatim from a story told by some cumulo-nimbus clouds as they floated over the chaparral foothills of California, one November afternoon. The human chorus recounts the tale while the computer voices assume the role of soloists who project the emotional content of the text – improbable voices of unthinkable animals who speak in a variety of unknown languages.

"Impossible Animals" was commissioned by the Hamilton College Choir. The recorded sound was created at the Center for Computer Research in Music and Acoustics (CCRMA) at Stanford University using a large computer system. The tape part, which is entirely synthetic, was created using the Chant synthesis program (Rodet) driven by the Pla music programming language (Schottstaedt) with extensions written by the composer.

"Il Nome" for Soprano and Tape (RICHARD KARPEN)

Commissioned by Judith Bettina.

Recent performances:
New Digital Music Concert, CCRMA Stanford University, January 1988.
Conference on Terrorism, Stanford University, February, 1988.

IL NOME for soprano and tape was composed in 1987. The text, "Il nome di Maria Fresu," is by the Italian poet Andrea Zanzotto. Maria Fresu was one of the 84 people killed in the August 2, 1980 bombing of the train station in Bologna, Italy attributed to the neo-fascist group Avanguardia Nazionale. She was blown up beyond any identification. The names of those killed form a memorial on the wall of the reconstructed train station. Along with the Zanzotto poem, the text for IL NOME includes a short passage from Monteverdi's "L'Orfeo." The Zanzotto text appears in fragments, not always in the correct order, until the end of the piece when the poem is sung through from beginning to end. The text from "L'Orfeo" occupies a section near the middle of the piece. IL NOME was composed for the soprano Judith Bettina whose voice is also the basis for much of the tape part. All of the vocal sounds on the tape (including the non-western chant-like singing) were derived from recordings of Ms. Bettina's voice. I also used recordings of breaking glass, a single note played on a violin and a single stroke of a tomtom. The recorded sounds as well as some purely synthetic sounds were processed in various ways using the Mixer program of Bill Schottstaedt. The piece has thousands of "splices" many of which were generated using the Pla language. The Samson Box was used for digital filtering and reverberation of the soundfiles. The effect I was aiming for in this piece was to make a tape part to which the live soprano part would seem naturally bound. As in other of my recent compositions involving the computer, I have relied on some techniques of "orchestration" which I have found useful in creating not only complex spectra, but a complex lively combination of sounds. One such technique is the use of a succession of similar sounds which overlap in time creating a much more dense "ensemble." Another technique makes use of very different sounds combined to create the perception of more complex single sounds.

"Exchange" for Flute and Computer-Generated Tape. (RICHARD KARPEN)

Commissioned by Laura Chislett.

Winner of 1987 Bregman Prize at the Bourges International Electroacoutic Music Contest, France.
Honorable Mentions at the 1987 International NEWCOMP Contest, Boston, and the Luigi Russolo Contest in Varese, Italy.
Recorded for Harmonia Mundi, Compact Disc, 1987 with the Groupe de Musique Experimentale de Bourges.
Scheduled for publication by Schott of West Germany in 1988.

Recent performances:
NEWCOMP, Boston, Mass., May, 1988.
SEAMUS Conference, Dartmouth College, New Hampshire, October, 1987.
Gaudeamus International Music Week, Amsterdam, September, 1987.
International Music Festival, Bourges, France, June, 1987.
Conference on the Humanities, Stanford, CA, April, 1987.

The primary synthesis technique for the realization for the tape part of EXCHANGE was the digital filtering of sounds having broad spectra. The synthesis and filtering was done entirely on the Systems Concepts Digital Synthesizer. An array of parallel filters (as many as 25 at a time) with narrow bandwidths and dynamically changing center frequencies was used to create the basis of the spectral as well as the harmonic, and in part, the melodic content of the work. Since the higher frequencies in the tape part are partials of lower fundamental frequencies, there is a direct relationship between the low pitched and high pitched material on the tape via the harmonic series.

The note lists for EXCHANGE were generated through algorithms written by the composer in the Pla Language. The flute material was also taken in part from the output of these algorithms. The flute is also required to play harmonics and multiphonics, adding to the effect already present in the tape part.

"Eclipse" for Computer-Generated Tape. (RICHARD KARPEN)

Winner of First Prize at the 1986 International NEWCOMP Contest.
Honorable Mention at the 1987 Bourges International Electroacoustic Music Contest.

Recent performances:
ICMC '87, University of Illinois, August 1987.
Luneburg, West Germany, March, 1987.
New Music Los Angeles Festival, L.A., Ca. March, 1987.
Intermusica Ensemble, Padua, Italy, November, 1986.
San Francisco Contemporary Music Players, November, 1986.
NEWCOMP, Boston, Mass., October, 1986.

ECLIPSE reflects my ongoing interest in evolutionary processes as models for musical structures. Several musical ideas evolve simultaneously, but their evolution also reflects their interaction with each other. There are areas of predominance for the different musical entities and periods during which each is dormant. The large-scale form is in two halves of almost equal duration. The first half is itself in two parts, during each of which a different type of musical element is predominant. In the second half of ECLIPSE there are more struggles for predominance among the musical materials. The pitch material for much of the piece (especially the second half) was generated through a procedure by which a "source" melodic idea is expanded by recursively interpolating copies of itself into the original. Each expansion becomes a new source and is expanded in the same way, creating ever longer streams of pitches. The programs to generate these pitch streams were written by the composer in the Pla Language. The emphasis for the creation of the sound materials for ECLIPSE is on "orchestration" rather than on the creation of self-sufficient sound entities. The synthesis techniques used include several types of Frequency Modulation, Amplitude Modulation and Additive Synthesis, but the primary "orchestration" technique involves the overlap of many iterations of "notes" to create complex dynamically changing sounds and musical textures. ECLIPSE is a "symphonic" work for computer. This is not to say that I have tried to imitate the sound of an orchestra. The piece is, rather, symphonic in its gestures, its length and in the general qualities of its musical textures. The sound of the piece itself should imply the presence of large forces. ECLIPSE was realized on the Systems Concepts Digital Synthesizer.

"Thus Spake Bosch" (STANISLAW KRUPOWICZ)

Performed in Warsaw, Stanford, San Francisco, Boston, Getenborg, Vancouver, Argentina, Paris, Szczecin (Poland), Katowice (Poland).
Awarded the Second Prize at the NEWCOMP International Competition in 1985.

"Multiplicity of Unity" or "Unity of Multiplicity" are pure illusions and mystifications. Nothing implies nothing, nothing derives from nothing, there are no causes nor effects. There is also no world of ideas, monads nor essence, therefore each reduction—even eidetic—is just a replacement of objects, or better, words which seem to be especially developed for this useless game. Everything may precede everything, everything may follow everything, and there is nothing which might not coexist with anything else. The ordering function of time is another illusion and oversimplification. The same applies to necessity, therefore "impossibility" is a common thing.

"Farewell Variations on the Theme by Mozart" (STANISLAW KRUPOWICZ)

There are 2 version of the piece: for tape only and for amplified string quartet and tape.
Performed at Stanford (tape version) in December, 1986 and Warsaw Autumn (version with string quartet) in September, 1987.
Awarded prize at the 8th Irino Prize Competition for Chamber Music in Tokyo, June 1987.

"... no, it's not a joke, Sir. It's not a joke as it's not cinematic music from a sinister movie. It's my imperfect tribute to the perfect composer, who was capable of seeing things as they are: neither funny nor tragic, or if you prefer, funny and tragic at the same time. It is my tribute to his wisdom which—yes, I must admit—is difficult to attain.

I always come back to Mozart when this trivial dialectic is too apparent in my life. This time I did the same. It helped me. It made me understand that when everything falls apart, there is always Kyrie Eleison left.

You are not serious, Sir, are you? You are not seriously serious? I like Mozart. I do like Mozart. And he was not.

No, it's not a joke. It's not cinematic music, either. It is a piece of my music, it's a piece of his music. And...don't be too serious; it ain't gonna work, Sir. It ain't gonna work..."

—Excerpts from an unwritten letter.

Special thanks to Toni Milosz from Peak Design for his extremely competent help with inserting sampled sounds and remixing the whole composition.

"Fantasmas de dos mundos" (SERVIO MARIN)

A music-theater piece for instruments, voice, dancers, actors, sculpture, lights and computer generated tape.

Performed at Stanford, May 1987.

This is a musical theater piece integrating instruments, voice, dancers, actors, lights, computer music, sculpture, and video. This kind of music, called by composer-director Servio Marin "Visonual Music" (1981), is a study of interrelations between movement, sound, light and visual elements.

The title "Fantasmas de dos mundos" comes from the work of Venezuelan writer Arturo Uslar Pietri who refers to Latin American culture as "the result of both the encounter and the struggle between European, American Indian and African cultures". In my piece, in order to emphasize this idea I have chosen the traditional form of variations (thematic and serial) as the European element of the piece. I also use the "Joropo" (Venezuelan folk music which is already a product of Spanish, Indian and African influences) as the raw thematic material. Finally I integrate these two worlds by using most of the electronic music techniques which have been developed in the last 30 years.

The atmosphere of the piece Ghosts, created from confrontations between reality and dreams, and from creative encounters between the past the present and the future are the guest of the party. They came from different pasts, times and spaces through ancestral and contemporary heritage. They apparently eclipsed the identity of their involuntary host who does not have a name nor a fate but whose goal is to understand the visitors. In this intemperate party humor, incongruities, wit, sounds, personages, movements become a parody of expectation.

This work was realized at CCRMA using the "Samson Box" and a Yamaha DX7 II-FD. It uses the serial transformations of the third minor interval as the "unifier element" (Shoenberg) of four musical events which intermix throughout the piece. These musical events, however, offer clear differences in timber and in style so that they can be heard as being different musical pieces. Internally, within the domain of intervals, the idea of "encounter" is given by the "degree of similarity provided by the serie" (a very European-Germanic concept borrowed from Max Deutsch). Externally we perceive the idea of struggle which is given by the contrast in mood, timber, musical style and rhythm. (characteristic of the Joropo). (A copy of the video tape of Fantasmas de dos Mundos is available from the composer.)

"Invisible Cities" (MICHAEL MCNABB)

Ballet music for piano, saxophone, computer tape, and electronics

Performances:
1987 AES Music and Digital Technology Conference
1986 Zellerbach Hall, Berkeley CA, with ODC/San Francisco Dance Co.
1986 16th International Experimental Music Festival, Bourges
1986 California State University, Long Beach (American Music Week)
1986 Dinklespiel Auditorium, Stanford University
premiere: December 6,7, 1985 Memorial Auditorium, Stanford University, with ODC/San Francisco Dance Co.

"Invisible Cities" was born from discussions between Michael McNabb and Gayle Curtis starting in 1984. Inspired by choreography which had been set to some of his earlier music, McNabb's desire was to compose a large form dance work, with several related movements around a common theme. At the same time, Curtis was participating in the work being done in machine choreography at the V.A. Robotic Aid project by Margo Apostolos and others, and wanted to see the concept carried further.

The idea of a joint effort was received enthusiastically by John Chowning, the director of CCRMA, who offered CCRMA's support and invited Brenda Way to consider participation by ODC/SF. The award of a National Endowment for the Arts grant to McNabb for the music, and the support of the Lively Arts at Stanford for the production allowed the collaboration to begin in earnest.

Inspired by choreography which had been set to some of my earlier music, I decided over two years ago to compose a large form dance work, with several related movements around a common theme. Although the bulk of the work would be computer synthesis, I also wanted to be able to perform in the work myself. This was partly a personal challenge, since I had been away from performing for awhile, and partly a way to try to directly raise the standards of music in dance performance, which I saw as too often neglected or weakly presented.

The novel "Invisible Cities" has always been one of my favorite books, and its beauty, concise structure, and dream-like imagery led me to consider it as the inspiration for my music, and to suggest it as the basis for the collaborative work. Appropriately, the music contains both subtle and explicit elements of various ethnic, popular or classical musics from different parts of the world, as well as sections of pure fantasy and musical imagery. It is more concerned with conveying the feelings and moods of the novel's settings than its intellectual concepts, leaving the latter in the capable hands of the choreographer and designer.

This music was funded by a grant from the National Endowment for the Arts. I would also like to thank CCRMA, its staff, and others there whose help in this effort was invaluable.

"Darkening" (IRA J. MOWITZ)

Performances:
New Digital Music, C.C.R.M.A., Stanford University, January 14, 1988.
Roosevelt Computer Music Festival, Roosevelt, NJ. , November 14, 1987.
American Music Week, San Jose State University, November 5, 1987.
S.E.A.M.U.S. National Convenion, Dartmouth College, October 22, 1987.

Originally "Darkening" was to have been the first part of a much larger piece, but in the process of assembling it, it became increasingly apparent that this opening music called out for a life of its own. (The remaining music is now a companion piece, "Shimmering.") It is just such turns of events that I have found most interesting about working in the computer music medium. The chance that what one sets out to do will in the course of the work become something else entirely—an end a beginning, a beginning an end, a small detail the main focus of a piece. There is something about the medium that suggests a constant reinvention of musical context—based not on what one planned on being able to realize before hearing anything, but on a continuous process of listening and responding to what one actually gets. Moreover, "Darkening" is based not on a clear, narrative notion of musical continuity, but instead proposes a less defined path, flowing through washes of sound and moods.

"Jubilum" (IRA J. MOWITZ)

Performances:
Muzyka Centrum Artistic Society, Teatra Bagatela, Krakow, Poland, March 7, 1988.
Skinnskatteberg Festival, Sweden, May 31, 1987.
WNYC broacast, July 26, 1987.
California Institute of the Arts, March 24, 1987.
Concert of the Berkeley Electronic Music Group (BECMUG) Bechtel Engineering Center, Sibley Auditorium, U.C. Berkeley, March 19, 1987.
Stanford University, December 11, 1986.
International Computer Music Conference, The Hague, Netherlands, October 1986.
Buenos Aires, Argentina, September 25, 1986.
Princeton University, May 19, 1986.

"Jubilum" is a celebratory piece, a journey through a wide range and many shades and kinds of celebration. The piece is clearly orchestral in conception, gesture and tone. The sounds aren't meant so much to be imitations of instrumental and orchestral timbres as they are to be evocative of that world. The piece was selected as one of the U.S. League I.S.C.M.'s six official submissions—and the sole computer piece—to the World Music Days 1987 held in Koln.

Both "Darkening" and "Jubilum" were realised on the Systems Concepts Digital Synthesizer (the "Sambox") at the Center for Computer Research in Music and Acoustics, Stanford University.

"Images" (ADOLFO NUÑEZ)

For voice, horn, trumpet, trombone, piano and computer-generated and processed tape.

Performances:
Stanford, July 1986
International Computer Music Conference, Urbana-Champaigne Illinois, August 1987.

IMAGES is based on the results of research into the perception of rhythm. For example, distortion of a composite rhythm at the beginning of the piece occurs because of unavoidable inaccuracies in the synchronization between performers. Yet another distortion occurs at the end of the piece when the performers are required to play with the tape in a rhythm having a period longer than the upper perceptible limit of synchronization, approximately 1.8 seconds. The soprano does not sing any texts, but rather creates timbre from phonemes.

The tape part, synthesized at CCRMA, requires that the performers follow it precisely in order to achieve a close dialog with it. Among other timbres, the tape uses simulations (images) of each of the five live instruments. The tape canonically mimics the performers' rhythms, sometimes mechanically, sometimes with human-like inaccuracy. Computer-assisted composition was used most extensively in the work's middle section. Pitches and rhythms were selected for the instruments and the tape by a program from lists which evolved according to a predetermined formal plan.

"Press" (ADOLFO NUÑEZ)

Recorded in LP by Circulo de Bellas Artes with the support of AT&T.

"Press" was realized at CCRMA using the "Samson Box" and programs developed by Bill Schottsteadt. All the timbres are synthetic. Some are simulations of acoustic instruments like marimba (realized by Xavier Serra), plucked strings (David Jaffe) and percussion (Jan Mattox) and other timbres are imagined (programmed by the author using FM).

"Press" is a divertimento on the perception of rhythm. A periodic rhythm (with a single duration) is transformed into another one with practically imperceptible irregularities. This ambiguous rhythm is transformed into another with four clearly different durations. The rhythm of the sun, even though very accelerated, is also present in this work. One curve that represents the activity of the sunspots from 1940 to 1982, and that like in other phenomena of nature is self-similar (fractal), is used from time to time to control the variation of intensity and timbre through time.

"JubJub" (HEINRICH TAUBE)

Recent performances:
1987 Lueneburg Germany
1987 London Sinfornietta
1986 Tanglewood Summer Music Festival
premier: Stanford, July 1985.

The composer wishes to express his thanks to Michael McNabb for allowing the use of his additive synthesis voice instrument, and to Bill Schottstaedt for both his help and his wonderfully adaptable FM violin instrument, which generated most of the sounds in this composition.

"Tremens" (HEINRICH TAUBE)

Recent performances:
1987 International Computer Music Conference at U. of Illinois
1987 Society for Electro-Acoustic Music in the United States, National Conference at Dartmouth
1987 Stanford

The title of the composition is taken from a line by Cicero, "toto pectorum tremens" ("the whole breast trembling"). The main musical ideas in this piece are all related to the act of trembling as a response to some basic emotional state such as anticipation, fear, panic, exhaustion, etc. In order to expand the timbral palette of the pitched material, I have incorporated into the composition synthetic, unpitched sounds such as wind, breathing, and large, struck sheets of metal. During the course of the work the Bach choral melody "Was bist du doch, o Seele, so betrubet" appears; the harmonization is mine and the choral setting is intentionally left incomplete. This composition was realized at CCRMA under an "artist-in-residence" grant from the Rockefeller Foundation. The composer would like to acknowledge Bill Schottstaedt for the use of his FM instrument, which was used to generate most of the sounds in this piece.

"Voices" (BRIAN SCHOBER)

Premiered at Stanford, July, 1985

"Voices" As the title implies, VOICES is entirely based on the human voice. The sonic material ranges from pure vocal sounds to highly processed and simulated vocal sounds. The musical material ranges from random vocal improvisation to meticulously notated pitch successions, often used simultaneously. The composer wishes to acknowledge Pat Mikishka, Harlan Hokin, and Clive Buckler for having provided much of the vocal material as well as Jan Mattox for her technical assistance.

"Sonata" (WILLIAM SCHOTTSTAEDT)

Premiered at Stanford, July, 1985.

"Sonata" was written in April and May, 1985. I originally wanted to synthesize the sound of a factory of some kind, but each metallic clang consumed most of the synthesizer, and I didn't have the energy to try to mix thousands of sample data files. The only obvious instrument is the FM violin, as usual. The piece is not cast in sonata form—I just couldn't think of an appropriate title.

"Put on a Happy Face (Water Music 2)" (WILLIAM SCHOTTSTAEDT)

Performed at Stanford Dec. 11, 1986.
Performed at Bourges Electroacoustic Music Festival, 1986.
Recorded on CCRMA Compact Disk Dec. 1986.

"Put on a Happy Face" is the second movement of "Water Music." It is based on two or three basic ideas, each of which is presented sped up or slowed down by as much as a factor of 60.

"Water Music 3" (WILLIAM SCHOTTSTAEDT)

Performances:
Japan, April 1987
Buenos Aires, Sept. 1987
Northwestern University, April 1987
Bourges Electroacoustic Music Festival, June 1987
Tanglewood, August 1987
Madrid, November 1987
San Francisco, December 1987
Pacific Rim Festival, San Diego 1987
Roro Festival, Sweden
Luenberg,Germany

"Water Music 3" is a complete revision of an earlier piece named "Sonata." My original intent was to use metallic clangs reminiscent of factory sounds, but these sounds are not entirely trivial to synthesize. The method used here is a combination of frequency modulation and additive synthesis.

"Brand X (Water Music 4)" (WILLIAM SCHOTTSTAEDT)

Performed at Stanford January 1988.
Performed in Argentina February 1988.

"Brand X Music" (or "California Music") is the last movement of "Water Music." It makes extensive use of Pythagorean and 11-tone tuning schemes.

"Leviathan" (WILLIAM SCHOTTSTAEDT)

Performances:
CCRMA May 27, 1987 (Industrial Affiliates concert).
Stanford July 16, 1987 (Frost).
Hanover, Germany, Aug. 1987 (as part of Cocteau's play "The Human Voice").

Leviathan is a pure concrete piece written during the spring of 1987. The original sounds included a dog howling, a horse neighing, a train, a pump, and a rooster. These were mixed, enveloped, filtered, resampled, and so on until they were scarcely recognizable. According to my calculations the final version of "Leviathan" involved 15220 "splices." The impetus for this came from one of the sea-stories of Melville ("Whitejacket" I think) in which a church bell gets loose during a storm and panics the sailors because they cannot parse the sound.

"Ladders and Plains" (AMNON WOLMAN)

Computer-generated stereo tape, for multimedia concert work produced by Jonathan Blum.
Premiered in WestBeth Theatre Center in New York, Febuary 1988.

The piece revolves around an upward motion (a ladder) complemented by a static motion (a plain). It was influenced indirectly by the poem "Murasaki Said" by the poet Mordechai Avi-Shaul. Here is a translation of the poem:

Murasaki Said (To Hagar)

Ah, dew drops
pray linger
on this flowering bud
so we can perceive
the form
it means to wear
when it blossoms.

(from Hebrew, A.W.)

"Practice (Attack of the Killer Keyboards)" (AMNON WOLMAN)

For clown, two pianos, harpsichord, fortepiano, organ, celeste, toy piano, DX7 (can be played on eight DX7's).
Commissioned by Tim Zerlang and the Stanford Music Guild.
Premiered on Febuary 1988, at Stanford University.

"Practice," is a set of variations based on the D major Musette by J.S. Bach. It was commissioned for the annual Pied Piper concert at Stanford, which this year was dedicated to the keyboard. The piece tries to educate the children about keyboards and their stylistic connotations.

"Medea" (AMNON WOLMAN)

For computer-generated stereophonic tape and four pre-recorded voices.
Finalist in the first Prix Ars Electronica 87, Austria.

The concept of MEDEA evolved while I was working on the music for the production of "Medea-Plays" conceived and directed by Ed Isser at the Stanford Drama department. Ed had the ingenious idea to use four narrators, backstage, each reciting a different version of the Medea story in a different language (Greek, Latin, French or German), while four actors on the center stage spoke in English. The rich musical possibilities that this idea offered attracted me immediately. With Ed's permission, I recorded the four narrators in the four languages. They spoke the text just as in the play, but I also included some variations such as having them whisper the text, or sing it on certain prespecified pitches. I then processed those recordings with a Lexicon digital reverberator, and the F4 computer, and combined them with some of the computer-generated sounds. In a sense, I was trying to juxtapose the Greek idea of child sacrifice as the ultimate personal tragedy, with the Judeo-Christian ideal of child-sacrifice as the ultimate religious experience, for example, Jesus and Isaac. MEDEA is dedicated to the memory of sacrificed childern.

"And then She said" (AMNON WOLMAN)

For actress, computer-generated stereophonic tape, four pre-recorded voices, and live computer graphics.
Premiered by Terra Vandergaw at the Stanford Humanities Conference, April 24th 1987.

AND THEN SHE SAID is a theatre-music piece for actress and computer-generated and processed sounds. It was realized using the System Concepts Digital Synthesizer, and the Lexicon digital reverbarator, at the Center for Computer Research in Music and Accoustics (CCRMA) at Stanford University. The piece was premiered at the Conference on Humans Animals, and Machines, organized by the Humanities Center at Stanford University, on Friday April 24th, 1987. The piece was videotaped the next day in a studio. About the piece the composer writes:

The concept of AND THEN SHE SAID evolved while I was working on the music for the production of "Medea-Plays" conceived and directed by Ed Isser at the Stanford Drama department. Ed had the ingenious idea to use four narrators, backstage, each reciting a different version of the Medea story in a different language (Greek, Latin, French or German), while four actors on the center stage spoke in English. The rich musical possibilities that this idea offered attracted me immediately. With Ed's permission, I recorded the four narrators in the four languages. They spoke the text just as in the play, but I also included some variations such as having them whisper the text, or sing it on certain prespecified pitches. I then processed those recordings with a Lexicon digital reverberator, and combined them with some of the computer-generated sounds I had used for the original Medea score.

The actress does not play the role of Medea; instead I chose to juxtapose the Greek myth and an opposite idea: the Judeo-Christian symbolism of child sacrifice as the ultimate religious commitment, as in the stories of Isaac and Jesus. The story that I adapted for this piece is that of Hanna and her seven sons, taken from the book of Maccabees.

An additional dimension is given to the piece by the accompanying computer graphics, which are controlled live during the performance. David Zicarelli wrote the Apple Macintosh program for creating these images; together we selected the images that would be most fitting for this piece. The mouse is used in real time to control the speed of the changes from one image to the next.

The piece came alive thanks to the help of my two friends Terra Vandergaw and Ed Isser. We started out with the music, the text, and some notions about the performance; they have made it into a coherent piece.

AND THEN SHE SAID was written between September 1986 and March 1987. It is dedicated to the memory of Catholic archbishop Oscar Romero of El Salvador, who was assassinated there on March 24, 1980 for his beliefs.

I would like to thank Sabine von Dirke, Beatrice Philibert, Martha Taylor, and Livia Tenzer, whose voices I used on the tape. Special thanks to John Chowning and Patte Wood for their support.

"M" (AMNON WOLMAN)

For orchestra and computer-processed stereophonic tape.
Premiered by the Stanford Chamber Orchestra, May 1986.

I worked with Marilyn Monroe's voice as sound material, with the idea that it would be fun to write a"happy piece". It turned out to be not as easy as I'd hoped. The desperateness of Marilyn Monroe came through. There were a lot of levels of dealing with that material and working out the piece. Most are by now too hard to describe, but perhaps the most important were the memories; the movie"M" by Fritz Lang, Andy Warhol's "Marilyn" series, and Charles Dodge's "Any resemblance is Purely Coincidental", were points of departure for this piece. Most of the tape part consists of transformations of Marilyn Monroe's singing taken from the movie "Some like it Hot", directed by Billy Wilder. These excerpts were manipulated digitally on the F4 computer at the Center for Research in Music and Acoustics (CCRMA) at Stanford University. The orchestra, while a separate entity, comments on or ignores the tape. Two literary quotes reflect different levels of the piece.

The first is by Jean Genet, from "Funeral Rites" (1969). "..The beauty of the lily lies similarily in the amazing fragility of the little hood of pollen that trembles at the top of the pistil. A gust of wind, a clumsy finger, a leaf, can break and destroy the delicate equilibrium that holds beauty in balance.

The second is by Kurt Vonnegut from "Dead-eye Dick" (1982). "..She told me, of walking into the music room of her father's mansion, which she had believed to be so indestructible when she was a little girl, and seeing what looked like foam, boiling out of the floor and a baseboard near the grand piano, and out of the legs and the keyboard of the piano itself. 'There were billions and billions of bugs with shiny wings, acting for all the world like liquid' she said.... .. nobody had played the piano for years. If somebody had played it, maybe it would have driven the bugs out of there. Father gave a piano leg a little kick, and it crumpled like it was made out of cardboard. The piano fell down."

EDUCATION

CCRMA is an interdisciplinary center administered by the Music Department at Stanford University. Classes and seminars taught at the center are open to registered Stanford students. The facility is also available to registered Stanford students for research projects which coincide with ongoing work at the center. Prospective graduate students should apply to the degree program at Stanford most closely aligned with their specific field of interest, i.e. music theory or composition, computer science, electrical engineering, psychology, etc., and should check with both CCRMA and their major department to make sure that there is a faculty advisor available to oversee the kind of work the student wishes to pursue. Acceptance in music theory or composition is largely based upon musical criteria, not knowledge of computing.

Admission requirements for degree programs can be obtained directly from each department.

CCRMA also offers summer workshops which are open to anyone wishing to apply.

Academic year curriculum:

MUSIC 120 INSTRODUCTION TO MUSIC SYNTHESIS AND PROGRAMMING USING MIDI-BASED SYSTEMS. (C. CHAFE)

Composition projects demonstrate participants' own software for voicing and controlling MIDI synthesis.

MUSIC 154 INTRODUCTION TO COMPUTER MUSIC (C. CHAFE)

Survey of recent compositions and computer based techniques used in music.

MUSIC 220 COMPUTER-GENERATED MUSIC

220A Fundamentals of Computer-Generated Sound (J. CHOWNING)

Introduction to computer-sound generation, composition, acoustics, and computer programming.

220B Compositional Algorithms, Psychoacoustics, and Spatial Processing (J. CHOWNING)

Use of a high-level programming language as a compositional aid in creating musical structures. Studies in the physical correlates to auditory perception, and review of psychoacoustics literature. Simulation of reverberant space and control of the position of the sound within the space.

220C Research (J. CHOWNING)

Research projects in composition, psychoacoustics, or signal processing.

220D Music Typography on the Computer. (L. SMITH)

MUSIC 320 SEMINAR IN SIGNAL PROCESSING FOR MUSIC RESEARCH (J. SMITH)

320A The Discrete Fourier Transform

Fundamentals of spectrum analysis for discrete-time signals. Topics: sampled signals, complex variables, geometric projection, orthogonality, the DFT, and Fourier theorems relating to time-shift, convolution, correlation, aliasing, signal power, symmetries, and bandlimited interpolation.

320B Applications of the Fast Fourier Transform (FFT)

Spectrum analysis and digital filtering using the FFT. Topics: convolution, recursive and nonrecursive filter structures, z-transforms, transfer function analysis, frequency response, time-varying filtering, choice of DFT windows, and use of the DFT to implement nonrecursive filters by means of the overlap-add or filter-bank summation techniques.

320C Recursive Digital Filtering

Analysis, design, and implementation of recursive digital filters. Concepts: difference equations, impulse response, transfer function, frequency response, poles and zeros, stability, phase and group delay, partial fraction expansion. Schur algorithm, physical simulation, and structural/numerical issues.

MUSIC 420 ADVANCED SEMINAR IN SIGNAL PROCESSING FOR MUSIC RESEARCH

420A Acoustical Signal Processing

Classical acoustics translated into a digital signal processing framework. Topics: mass-spring oscillation, the mass-spring chain, the wave equation for the ideal flexible string and acoustic tubes, traveling waves, wave impedance, scattering theory, signal energy and momentum, digital filter counterparts, allpass techniques, and efficient physical modeling using delay lines, scattering junctions, and low-order digital filters.

420B Nonlinear Modeling

Computational models for woodwinds and strings. Physically meaningful synthesis algorithms are built by coupling a "negative-resistance device" (woodwind reed or bow-string interaction) to a linear filter (which models a woodwind bore or vibrating string). The models are designed to capture only the "audible physics" of a musical instrument with a computationally simple algorithm.

420C Linear modeling

Techniques for system identification and linear prediction. Computational methods are described for designing digital filters which automatically adjust free parameters to match physical measurements of linear resonating components of musical instruments. A special case is linear predictive modeling of speech.

Special workshops during the Summer:

SYNTHESIS AND COMPOSITION ON SMALL ADVANCED SYSTEMS (C. CHAFE, J. CHOWNING)

The course includes a detailed exploration of the theory and programming of the YAMAHA "X" series digital synthesizers of which the DX7 will serve as the basis of instruction. Discussion will include computer based modular systems having MIDI inter-connections into which the digital synthesizers can be effectively integrated. Lectures on digital synthesis, Frequency Modulation theory, DX7 programming, relevant acoustics and psychoacoustics, and composition using small computers are given on weekday mornings with "hands-on" application sessions in the afternoons. The course is designed for composers and keyboard players who have had some previous experience with an "X" series synthesizer.

SYNTHESIS AND COMPOSITION ON A REAL-TIME DIGITAL SYNTHESIZER PROCESSOR (W. SCHOTTSTAEDT)

This course is based on the use of CCRMA's unique digital synthesizer-processor, designed by Peter Samson and of the compositional program Pla, written by Bill Schottstaedt. Lectures on digital synthesis, Pla, acoustics and psychoacoustics are given on weekday mornings with free access to the CCRMA system at other times. The course is designed for composers who wish to complete a two- or four-channel computer-synthesized piece which is recorded onto tape.

MUSIC PRINTING ON SMALL COMPUTERS (L. SMITH)

This workshop concentrates on the use of the MS computerized music printing system which has been developed at Stanford over a period of more than ten years and has produced a wide variety of musical publications. In addition to the main instruction, lectures are given on problems in real-time computer graphics and the history and practice of music typography. This workshop is designed for composers who wish to be able to present their music in copy that equals the quality of professional publication. Also attention is given to musical scholars who might wish to produce a body of musical examples for their dissertations, etc. Small graphics systems similar to the IBM PC are used. Each student should be able to produce several pages of high-quality hard copy during the session.

ACTIVITIES

Regular activities at CCRMA include monthly demonstrations of ongoing work open to the public and regularly scheduled seminars under the title of "Odd Thursdays." Topics presented at Odd Thursdays are related to work of interest to the CCRMA community. Odd Thursday presentations during the academic year 1986-1987 included talks by Marc LeBrun of Symbolics, Inc.; Albert Bregman, Visiting Professor of Psychology from McGill University; Stephen Pope, Research Scientist from Xerox PARC; John Stautner of Compusonics; Jim Angell, Professor of EE at Stanford; Vincent Salmon; Barry Vercoe, Professor at MIT Media Lab; Diana Deutsch, Professor of Psychology, UC San Diego; Richard Lyon, Researcher at Schlumberger Palo Alto Research; and Mitch Weintraub. Presentations were also made by National SemiConductor and IMS, Inc.

Special presentation/concerts were also given by Gordon Mumma, composer; Steve Reich, composer; Stefano Scodanibbio, double bass player and composer from Italy; Laura Chislett, flutist and expert in extended flute techniques from Australia; Marek Kielczewski, composer from Krakow, Poland; Johannes Goebel, composer and builder of instruments from Hannover, Germany; and Gerhard Staebler, composer and organist from Essen, Germany.

The First Annual CCRMA Associates Meeting was held at CCRMA at the end of May, 1987. Member companies represented included Apple Computer, Dynacord, Roland Japan and U.S., Sequential Circuits, Symbolics, Xerox PARC, and Yamaha. The three-day seminar included demonstrations, presentations of ongoing work by research staff and graduate students, a dinner and concert with guest speaker John R. Pierce, a banquet dinner with guest speaker Pete Samson, and informal discussions.

Other activities included presentations to the Knights Fellows program at Stanford; the Computer Science Forum; the Humanities Center Conference on Humans, Animals, Machines: Boundaries and Projections; the Audio Engineering Society; the Music Library Association; the College Music Society; the Society for Motion Picture and Television Engineers; the Stanford Club of Palo Alto; and the Stanford Mothers Club. The Music Guild at Stanford sponsored a special workshop on computer music for local high school students.

VISITORS

Many people come to CCRMA to visit the center, talk with people, give lectures, get information, etc. Here is a representative list of some the visitors during the past two years.

Gerhard Staebler, composer from Germany
Jean-Francois Allouis, IRCAM
Milton Babbitt, composer, Princeton
Nikolai Badinski, composer
Jean-Baptiste Barriere, IRCAM
Leslie Bassett, composer, Michegan
Charles Boone, composer, San Francisco
Henry Brant, composer, Santa Barbara
John Seeley Brown, XEROX PARC
Don Buchla, Berkeley
Ellen Buchwalter, Rockefeller Foundation
Claude Cadoz, ACROE, Grenoble, France
Laura Chislett, flutist, Australia
Marek Cholonewski, Polish composer
Gayle Curtis, Veterans Administration Rehabilitation
Giovanni Debiasi, CSC Universita di Padua, Italy
Dianna Deutsch, USCD faculty
Marcella DuCray, harpist, San Francisco
Richard Felciano, composition faculty, UC Berkeley
Dave Finley, Floating Point Systems
Mr. Fujimora, Yahama
Adrien Freed, IRCAM and Droidworks
John Gatts, YIC, Buena Park
Gordon Getty, composer
Stan Getz, saxophonist
Michael Gordon, composer
Robert Gross, Lucasfilm
David Haynes, IMS Inc.
William Hewlett, CCRH
Karl Hirano, Yamaha, Japan
Anthony Holland, composer, Swarthmore
Heinz Holliger, oboist and composer, West Germany
Daniel Kolbialka, violinist, San Francisco Symphony
Stanley Jungleib, Sequential Circuits
Hiro Kato, Yamaha, Japan
Gary Kendall, Northwestern University
Mark Koenig, YIC, Buena Park
Marc LeBrun, Symbolics
Mark Lentczner, Apple Computer
Gary Leuenberger, Yamaha
D. Gareth Loy, CARL, UCSD
Annie Luciani, ACROE, Grenoble, France
Fred Malouf, Sequential Circuits
H. Matsuoka, Roland, Japan
F. Richard Moore, CARL, USCD
Jim Mothersbaugh, Roland, Los Angeles
Lyman Miller, Hewlett-Packard
Carl Muller, Xerox PARC

Gordon Mumma, composer, U.C. Santa Cruz
Mr. Nagai, Yamaha, Japan
David Oppenheim, Opcode Systems
Larry Oppenheim, composer
Arvo Paert, composer, Estonia, Russia
Vince Perry, Sony
Rudolf Rasch, psychoacoustician and musicologist
Steve Reich, composer
Dirk Reit, composer, West Germany
Roger Reynolds, composer and faculty, UCSD
Vince Salmon, acoustic consultant, San Francisco
Peter Samson, Systems Concepts
Andrew Schloss, faculty, Brown University
Gerald Schrutz, Dynacord
Stefano Scodanibbio, bassist, Italy
Peter Setz, Karlsrue, GA
Wayne Slawson, UC Davis
Brian Smith, Xerox PARC
Dave Smith, Sequential Circuits
Michael Stillman, Djerassi Foundation
Allen Strange, composer, San Jose State University
Hans Tchernig, Dynacord
Mark Tsuruta, Roland, Los Angeles
Barry Vercoe, MIT
Alejandro Vinao, composer, Argentina
Andy Voekel, Symbolics
Mr. Wang, Director Peking Opera
Mitch Weintraub, SRI
Alexander Wiyic, composer from Belgrade
David Wessel, IRCAM
H. Yasunaga, Roland, Japan

FACILITIES

CCRMA is one of the top-rated facilities for digital sound research in the world. The Center is located on the Stanford University campus in a building that has recently been refurbished to meet its unique needs. The facility includes a large computer room, a large quadraphonic experimental space with adjoining control room/studio, an all-digital recording studio with adjoining control room, a MIDI-based small systems studio, several work areas with terminals, personal computers, synthesizers and speakers, a seminar room, an in-house reference library, a classroom and offices. The building has been "hard-wired" so that any office or workspace can connect with the mainframe computer and synthesizer, Ethernet and AppleTalk. A network gateway connects to the campus at large and also to national and international networks. A description of the hardware and software environment follows below.

The CCRMA hardware environment consists of a mainframe computer and a network of workstations, including Symbolics LM2 LISP machines connected to an FPS-120 Array Processor, Xerox Dandy Tiger LISP Machines, AT&T PC's, and Apple Macintosh Plus's and Macintosh II's. The heart of the computing environment is the Foonly F4 mainframe which emulates a PDP-10. The Foonly has 4 Mbytes of memory, 1.5 Gbytes of disk storage and two magnetic tape drives. It runs the WAITS operating system, a legacy of CCRMA's years within the Stanford Artificial Intelligence Laboratory. WAITS provides time-sharing service, supporting 16 graphics terminals and several modems, and servicing a number of special-purpose DMA devices. Two of the most unique devices are the Systems Concepts Digital Synthesizer (aka "Samson Box") and the "Sonly Interface." The Samson Box is one of the most powerful audio signal processors in the world, capable of computing a large number of programmable digital oscillators, generators, filters and modifiers. Data from the Samson Box can be sent to its four DACs, or the samples may be written to disk for access. The "Sonly interface" allows the Foonly to communicate with the Sony DAP-1610 and PCM-F1 processors for digital audio transfer, recording, and AD/DA conversion. Another AD/DA facility is connected to a Macintosh II (Dyaxis System with 45 minutes of stereo storage capability at a 48kHz sampling rate). Most mainframe workstations are equipped with a stereo or quadraphonic analog audio system. Audio signals from any of 16 audio channels are routed to selected terminals by a computer-controlled switcher. Text and graphic hardcopy is provided by an Imagen/Canon Laser Printer, a Printronix Printer-Plotter and a large-format Versatec (8000 series) printer. Hardcopy can also be obtained via the Ethernet and an Apple Laserwriter.

The CCRMA software has been developed over a twenty-year period, and consists of a vast set of programs and system tools for editing, viewing, synthesizing, and analyzing sound. Much of the software was written in SAIL, a sophisticated Algol-like language currently maintained in-house, or PDP-10 assembly language for use on the Foonly. Two interpreted languages were developed at CCRMA. Pla is a high-level language designed for note-list processing. It combines LISP-like and SAIL-like features and includes parallelism and other musically useful constructs. SP is a signal-processing language that includes dynamic loading of the entire IEEE signal processing library. As part of the move toward distributive processing at CCRMA, translators have been written to convert from SAIL to C and from SAIL to LISP. Programs for managing huge FORTRAN libraries and generating foreign-language calling sequences have been written as well. Recent software has been written for the new personal computers and Lisp machines using InterLisp D, ZetaLisp, Le_Lisp, Smalltalk and C. An effort is underway to define a protocol to transport software among the different systems.

A large body of software exists for manipulating sound files, i.e., files containing sampled audio signals. These files can be graphically edited, analyzed, mixed and filtered. A number of other file formats exist, including "plot files" for graphic output, "merge files," which are bundles of sound files, and "filter files," which contain filter specifications. Programs for converting between all formats are provided. Available sound file analysis techniques include linear prediction and phase vocoder analysis. In addition, programs are available for designing filters by virtually all known methods.

The Samson Box, one of the world's most heavily used large audio signal processors, has spawned software for a wide variety of synthesis algorithms. The Samson Box software supports parallelism, automatic allocation, a message-passing modular environment, and high-level real-time interaction. Diagnostic programs provide sophisticated debugging capabilities, including breakpoints, readback, multiple representations, and analysis.

Software for high-level polyphonic music recognition from audio or MIDI sources is being developed on the Xerox Dandy Tigers and on the AT&T and Apple PC's. For audio signals, the "Bounded Q Transform," developed at CCRMA, provides multiple-rate phase-vocoder-like analysis, and gives a variety of time-resolution and frequency-resolution views of the same data. The growing emphasis on LISP facilitates the sharing of software with other centers and researchers. Recent software aquisitions include the SPIRE speech workbench software from MIT, the MACSYMA symbolic mathematics package, and MIDILisp.

Representative List of Equipment

Foonly F4 central processor with 1.5 Mwords memory, 1.5 Gbytes of disk storage, two magnetic tape drives, graphics processor with 16 terminals, modems, interface to the "Samson Box" and the "Sonly Interface", printers and plotters
2 Valid workstations
3 Xerox Dandylion workstations with printer
4 Symbolics Lisp machines, model LM2, with 300 mbytes disk drives, tape drive, color monitors, and interface to the FPS
7 Apple Macintosh Plus with MIDI interface
4 Apple Macintosh II's with color monitors and hard disks

Systems Concepts digital synthesizer/processor ("Samson Box")
FPS array processor
digital-to-analog converter 16 bits (custom made)
analog-to-digital converter 12 bits (custom made)
8 300 mbyte disk drives, 18 300 mbyte disk packs
1·670 mbyte Winchester disk drive
1 Printronics line printer
1 Versatec graphics printer
1 Imagen laser printer
1 "Sonly" interface to Sony PCM recorders
16 terminals w/keyboards
2 Apple Imagewriters
1 Apple Laserwriter
Ethernet

3 Yamaha DX7IIFD synthesizer/keyboards
2 Yamaha DX7II0 synthesizer/keyboards
5 Yamaha FBO1 digital synthesizers
2 SPX90II digital sound processor
1 MEP4 MIDI event processor
5 YRM103 DX7 voicing program
1 Prophet 3000 Sampler
1 IVL Pitch Tracker
12 pairs Sony headphones

Quadraphonic experimental workspace and dubbing facility

1 Scully 4channel 1/2 track tape machine
2 Sony 4channel 1/2 track tape machine
2 Ampex ATR 4 channel 1/2 track tape machines
2 Ampex ATR 4 channel 1/4 track tape machines
2 Sony cassette tape recorders
1 Yamaha 4 channel cassette tape recorder
4 Nakamichi cassette tape recorders
2 Yamaha P2100 amplifiers
4 Yamaha P2200 amplifiers
4 Yamaha 1030 cross-over networks
4 DBX and Dolby noise reduction units
1 Quantum 8 channel mixing console
1 Lexicon 244XL digital reverberator

4 Yamaha custom-made speakers
4 MDM4 speakers
1 video playback system
1 compact disk player

Digital Recording studio

1 IMS Dyaxis Digital processing system
2 Meyer 833 monitor speakers
4 EM Long TA2 loudspeakers
Stax headphones
4 microphone stands
2 Crown PZM mics
4 Schoeps BLM mics
1 IMS microphone preamp
1 Sony PCM-F1 digital audio processors
2 Sony PCM-701 digital audio processors
1 Sony SL-2000 video cassette recorders
1 Sony SL-2500 video cassette recorders
1 Yamaha C7 Grand Piano

Digital mixdown station

2 EM Long TA-2 loudspeakers
1 Sony PCM-701 Digital Audio processor
1 Sony PCM-F1 Digital Audio processor
2 Sony SL-2500 Video Cassette recorders
1 Sony SL-2000 Video Cassette recorder
Sony headphones

MIDI-Based Synthesizer Studio

1 Yamaha DX7II programmable algorithmic synthesizer
1 Yamaha KX88 remote keyboard
1 Yamaha TX816 FM Tone Generation System
1 Yamaha RX11 Digital Rhythm Programmer
1 Yamaha QX1 Digital Sequence Recorder
1 Yamaha CX 5MF Music computer
1 Yamaha RM2408 24 track Recording Mixer
4 Yamaha D1500 Digital Delays
2 Yamaha REV7 Digital Reverberators
1 Yamaha YME8 MIDI Expander
2 Yamaha P2200 amplifiers
4 Yamaha NS1000 Monitor speakers
2 Mac + Computer w/ MIDI Interface
1 Synclavier w/ peripherals
1 TEAC 8 track analog recorder

Portable concert system

4 Yamaha crossover networks
4 Yamaha P2200 amplifiers

1 Yamaha 16 channel mixer
2 DBX noise reduction units
4 road cases
4 Meyer MSL3 loudspeakers
4 Meyer 650R2 subwoofers
8 Ashly FET500 power amplifiers

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