CENTER FOR COMPUTER RESEARCH IN MUSIC AND ACOUSTICS
DEPARTMENT OF MUSIC, STANFORD UNIVERSITY
REPORT NO. STAN-M-104

CCRMA OVERVIEW

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1 General Information

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

Areas of ongoing interest at CCRMA include: Applications Hardware, Applications Software, Synthesis Techniques and Algorithms, Physical Modeling, Real-Time Controllers, Signal Processing, Digital Recording and Editing, Psychoacoustics and Musical Acoustics, Music Manuscripting by Computer, Composition, and Real-Time Applications.

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. Departments actively represented at CCRMA include Music, Electrical Engineering, Mechanical Engineering, and Psychology.

Center activities include academic courses, seminars, small interest group meetings, summer workshops and colloquia. Concerts of computer music are presented several times each year 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 periodically.

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, the Journal of the Acoustical Society of America, and various transactions of the Institute of Electrical and Electronic Engineers (IEEE). Compositions are presented in new music festivals and radio broadcasts throughout the world and have been recorded on cassette, LP, and compact disk.

CCRMA is affiliated with the Center for Computer Assisted Research in the Humanities (CCARH), also located at Stanford. CCARH conducts research on constructing computer databases for music, and on creating programs that allow researchers to access, analyze, print, and electronically perform the music. This focus is complementary to research at CCRMA in several ways.

Support for CCRMA has been received from the late Doreen B. Townsend, Walter Hewlett, the California Arts Council, the Ann and Gordon Getty Foundation, the Mellon Foundation, the National Endowment for the Arts, the National Science Foundation, the Rockefeller Foundation (for artists-in-residence), the System Development Foundation, Apple Computer, ATR Human Information Processing Research Labs, Aureal Semiconductor, Bio Control, Crystal Semiconductor, Digidesign, Dynacord, E-mu, Fast Mathematical Algorithms and Hardware, Interval Research, ITRI CCL Taiwan, Korg, Matsushita, Media Vision, NEC, NeXT Computer, NTT Basic Research Labs, Opcode Systems, Philips Semiconductors, Rockwell International, Roland, Symbolics, Texas Instruments, Xerox Palo Alto Research Center, Yamaha, Young Chang R&D Institute, Zeta Music Partners, and private gifts.

2 Roster

For the latest information on the denizens of CCRMA, see their individual home pages.[1] Below is a tabulation organized by group. The home page URL for each person is constructed from the login name as "http://www-ccrma.stanford.edu/~login".

2.1 Staff and Faculty

Login	Name	Position
cc	Chris Chafe	Associate Professor of Music, CCRMA Director
brg	Jonathan Berger	Associate Professor of Music
jos	Julius O. Smith III	Associate Professor, Music and (by courtesy) Electrical Engineering
mvm	Max V. Mathews	Professor of Music (Research)
jdh	Jonathan Harvey	Professor of Music
dbs	David Soley	Assistant Professor of Music
nando	Fernando Lopez-Lezcano	System Administrator / Lecturer
jay	Jay Kadis	Audio Engineer / Lecturer
hmk	Heidi Kugler	Secretary
bil	William Schottstaedt	Research Associate
gary	Gary Scavone	Research Associate
jc	John Chowning	Professor of Music, Emeritus
lcs	Leland Smith	Professor of Music, Emeritus
jrp	John R. Pierce	Visiting Professor of Music, Emeritus
esf	Eleanor Selfridge-Field	Consulting Professor of Music
n/a	Walter B. Hewlett	Consulting Professor of Music
mab	Marina Bosi	Consulting Professor of Music

[1] http://www-ccrma.stanford.edu/CCRMA/HomePages.html

2.2 Music Graduate Students

Login	Name	Degree Program
lonny	Lonny Chu	PhD Computer-Based Music Theory and Acoustics
pchordia	Parag Chordia	PhD Computer-Based Music Theory and Acoustics
tkunze	Tobias Kunze	PhD Computer-Based Music Theory and Acoustics
randal	Randal Leistikow	PhD Computer-Based Music Theory and Acoustics
unjung	Unjung Nam	PhD Computer-Based Music Theory and Acoustics
cnichols	Charles Nichols	PhD Computer-Based Music Theory and Acoustics
norton	Jonathan Norton	PhD Computer-Based Music Theory and Acoustics
sile	Sile O'Modhrain	PhD Computer-Based Music Theory and Acoustics
juan	Juan Pampin	PhD Computer-Based Music Theory and Acoustics
craig	Craig Sapp	PhD Computer-Based Music Theory and Acoustics
savd	Scott Van Duyne	PhD Computer-Based Music Theory and Acoustics
leigh	Leigh VanHandel	PhD CCRMA-cology (on leave 97-98)
pvh	Paul von Hippel	PhD Computer-Based Music Theory and Acoustics
fujishim	Takuya Fujishima	MA Science and Technology
issac	Issac Roth	MA Science and Technology
aguiar	Celso Aguiar	DMA Composition
oded	Oded Ben-Tal	DMA Composition
ching	Ching-Wen Chao	DMA Composition
jmd	Janet Dunbar	DMA Composition
falk	Chris Falk	DMA Composition
rjfleck	Robert Fleck	DMA Composition
nicky	Nicholas Hind	DMA Composition
junkim	Jun Kim	DMA Composition
senylee	Seungyon-Seny Lee	DMA Composition
bobby	Bobby Lombardi	DMA Composition
jmcginn	John McGinn	DMA Composition
kotoka	Kotoka Suzuki	DMA Composition

2.3 Engineering Graduate Students

Login	Name	Degree Program
bilbao	Stephan Bilbao	PhD Electrical Engineering (on leave 97-98)
dpberner	David P. Berners	PhD Electrical Engineering
scottl	Scott Levine	PhD Electrical Engineering (on leave 97-98)
vickylu	Hui-Ling Lu	PhD Electrical Engineering
putnam	William Putnam	PhD Electrical Engineering (on leave 1997)
stilti	Timothy Stilson	PhD Electrical Engineering
yoonie	Yoon Kim	PhD Electrical Engineering

2.4 Undergraduate Students

Login	Name	Degree Program
n/a	Jabari Anderson	Music, Science and Technology
n/a	Vincent Duron	Music, Science and Technology
corky	Corky Gainsford	Music, Science and Technology
daniel	Daniel Gould	Music, Science and Technology
pph	Patty Huang	Music, Science and Technology
beauty	Anndretta Lyle	Music, Science and Technology
blackrse	John A. Maurer, IV	Music, Science and Technology
n/a	Crystal McCreary	Music, Science and Technology
ajm	Adam Miller	Music, Science and Technology
aj	Andrew Nelson	Music, Science and Technology
gramps	John Niekrasz	Music, Science and Technology
kris	Christine Ohm	Music, Science and Technology
jaiscnce	JaiJae Soto	Music, Science and Technology
keach	Keach Hagey	Music, Science and Technology (minor)
aqueelah	Aqueelah Haqq	Music, Science and Technology (minor)
ericm	Frederick Miller	Music, Science and Technology (minor)
n/a	Scott Scruggs	Music, Science and Technology (minor)
entropy	Gabriel Serrano	Music, Science and Technology (minor)

2.5 Visiting Scholars

Login	Name	Home Affiliation
alford	Ron Alford	Composer, USA
jdc	Joanne Carey	Composer, USA
jan	Jan Chomyszyn	Psychoacoustic Researcher, Poland
mo	Maureen Chowning	Singer/Composer, USA
gius	Giuseppina Colicci	Ethno-Musicologist, Italy
fornari	Jose Fornari	PhD Student, University of Campinas, Brazil
jyri	Jyri Huopaniemi	Electrical Engineer, Helsinki Technical University, Finland
daj	David Jaffe	Composer, Software Research Engineer, USA
kurz	Michael Kurz	Electrical Engineer, Germany
peer	Peer Landa	Composer, Norway
eig	Enrique Moreno	Researcher, Mexico
muller	Carl Muller	Researcher, USA
ako	Akiko Orita	Media/Music, Japan
pasi	Fiammetta Pasi	Composer, Italy
n/a	Yasuhiro Takenaka	Composer, Japan
cello	Sean Varah	Researcher, Harvard University
patte	Patte Wood	ICMA, USA
marek	Marek Zoffaj	Composer, Slovakia

2.6 Collaborators

Login	Name	Affiliation
bau	Marcia Bauman	Lecturer, Music Department
prc	Perry R. Cook	Assistant Professor, Computer Science and Music, Princeton University
dhuron	David Huron	Researcher, Canada
dex	Dexter Morrill	Professor, Composition, Colgate University
levitin	Daniel Levitin	Lecturer, Music Department
xjs	Xavier Serra	IUA - Phonos, Universitat Pompeu Fabra, Barcelona, Spain
malcolm	Malcolm Slaney	Lecturer, CCRMA
hkt	Rick Taube	Assistant Professor, Composition, University of Illinois

2.7 Industrial Affiliates

Company	Address
Digidesign	Palo Alto, CA
E-mu Systems	Scotts Valley, CA
Fast Mathematical Algorithms and Hardware	Hamden, CT
Industrial Technology Research Institute	Hsinchu, Taiwan
Interval Research Corporation	Palo Alto, CA
NTT Basic Research Labs	Kanagawa, Japan
Opcode Systems	Palo Alto, CA
Roland Corporation	Osaka, Japan
Texas Instruments	Dallas, Texas
Yamaha Corporation	Hamamatsu-shi, Japan

3 Facilities

CCRMA is located on the Stanford University campus in a building that was refurbished in 1986 to meet its unique needs. The facility includes a large space with quadraphonic sound for teaching, concerts, and acoustic experimentation, an adjoining control room/studio, a multi-track recording studio with adjoining control room, two MIDI-based studios, several work areas with workstations, synthesizers and speakers, a seminar room, an in-house reference library, classrooms and offices. The building has been wired so that any office or workspace can connect with the workstation network. A gateway connects the workstations to the campus at large and also to the Internet. A description of the hardware and software environment follows below.

The CCRMA computing environment is supported by more than 40 machines that include NeXT workstations, Pentium and Pentium Pro PCs dual booting NEXTSTEP and Linux, Silicon Graphics workstations and Macintosh computers. All machines are connected through a switched fast ethernet backbone and several high speed servers provide shared services and resources to all computers in a way that is transparent to the users. An ethernet connection to the Stanford University Network (SUNET) provides high speed connectivity with the rest of the world. Soundfile manipulation and MIDI input and output are supported on all platforms. Four channel soundfile playback is supported on the NeXTs through custom built 4 channel D/A converters, on the Pro Tools systems, and on SGI workstations. Digital audio processors include a Studer-Editech Dyaxis II system, two Digidesign Pro-Tools system with a Sony CD-R drive, several Singular Solutions analog and digital audio input systems for the NeXTs, and several Panasonic DAT recorders. Text and graphics are handled by an HP 4c color scanner on the unix-based systems and by two NeXT Laser printers and an AppleLaserwriter 630 Pro.

The MIDI-based systems include various Macintosh computers with Yamaha, Roland and Korg equipment including Yamaha DX, SY, TG and VL synthesizers, KX88 keyboard controller, Korg X3R, I3, WaveDrum and WaveStations, an E-mu Systems Emulator IV and ESI-32 sampler, and digital delays and reverberation. The mixing console is a Mackie 24-8 in the MIDI Studio and an Allen and Heath GL2 in Studio E, while multitrack recording is supported by DA-38 digital 8-track recorders. Speakers are Ramsa WS-A200: stereo with subwoofer in MIDI studio and quad in Studio E. Other equipment available includes IVL pitch trackers, a Buchla Lightning MIDI controller, several Mathews Radio Drum controllers, two Opcode Studio 5 MIDI patchers, and drum machines from Yamaha.

Studio recording equipment includes a Biamp Legend 20x16 in-line console, a Yamaha DMR-8 8-track digital recorder/mixer with eight-channel 19-bit A/D converter, several Tascam DA-88 digital 8-track recorders, a Tascam 80-8 analog 8-track recorder with dbx noise reduction, Ampex 104 4-track and 800 series stereo analog recorders, Alesis ADAT, and a Panasonic SV-3700 DAT recorder. Outboard equipment includes Lexicon 224XL and Yamaha REV-7 reverbs, compressors including Teletronix LA-2A, 2 Behringer Composers and a dbx 166, Rane GE-30 equalizers, and effects processors including Korg A1, Yamaha SPX-1000 and SPX-90II and D-1500 delays. Monitor speakers are Westlake BBSM-10 powered by Hafler P-230 amplifiers and JBL 4206 powered by a QSC 1080 amplifier. CCRMA has an assortment of microphones including Neumann TLM-193, AKG C414B-ULS and C-460B, Sennheiser MD-421, Electrovoice RE-20, Crown PZM, and Shure SM- and Beta 57s. Other equipment including a Soundcraft 200 Delta console (eight mic/line and eight stereo line input modules), 4 Meyer MSL-3/650R2 speakers with Ashly FET-500 amps, a Tascam DA-P1 portable DAT recorder, and microphones from B&K (4006) and Schoeps (BLM3) are available for concert production and remote recording.

The digital editing studio (Studio D) consists of a Dyaxis II digital editing system, Digidesign Sound Designer II system, Meyer 833 monitor speakers, a Yamaha DMP-7D digital mixer, Yamaha SPX-1000 processor, a Panasonic SV-3700 DAT recorder and Macintosh and NeXT computers. Digital and analog patchbays provide interconnection.

A new digital editing/MIDI studio (Studio E) features a Digidesign 32 track Pro Tools III system with 36 DSP chips on 9 Farms, a 12-slot rackmount NuBus expansion chassis, SampleCell II sampler cards, an Adat interface, a SMPTE Slave Box, and an 888 I/O 8 channel audio interface.

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 originally written in SAIL, a sophisticated Algol-like language for use on the previous mainframe and has been ported to the new workstation environment and developed further. The programs currently in use include a comprehensive environment written in Common Lisp that includes Common Lisp Music (CLM) for music synthesis and signal processing, Common Music (CM) and STELLA for compositional programming and Common Music Notation (CMN) for creation of common music notation scores. The lisp-based world closely interacts (on the X windows environments) with Snd, a very complete sound editor and mixing tool also developed at CCRMA. Recent projects in music recognition, real-time performance, audio, signal processing, acoustics, psychoacoustics and physical modeling have been developed in languages native to the workstations, primarily Common Lisp, C, C++, Objective-C, Matlab, Mathematica, and Smalltalk. A graphical environment for real-time DSP research, SynthBuilder, is also used under NEXTSTEP. Of course there is a wide variety of public domain software for text, image and sound processing installed on all workstations.

4 Courses

CCRMA is a part of the Department of Music at Stanford University. Classes and seminars taught at the center are open to registered Stanford students and visiting scholars. The facility is also available to registered Stanford students and visiting scholars for research projects which coincide with ongoing work at the center.

Prospective graduate students especially interested in the work at CCRMA should apply to the degree program at Stanford most closely aligned with their specific field of study, e.g., Music, Computer Science, Electrical Engineering, Psychology, etc. Graduate degree programs offered in music are the MA in Music, Science, and Technology; the DMA in Composition; and the PhD in Computer-Based Music Theory and Acoustics. 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 particular department. CCRMA does not itself offer a degree.

The Music Department offers both an undergraduate major and minor in Music, Science, and Technology (MST). The MST specialization is designed for those students with a strong interest in the musical ramifications of rapidly evolving computer technology and digital audio and in the acoustic and psychoacoustic foundations of music. The program entails a substantial research project under faculty guidance and makes use of the highly multi-disciplinary environment at CCRMA. This program can serve as a complementary major to students in the sciences and engineering. Requirements for the undergraduate programs are available from the Stanford Music Department.

4.1 University Courses at CCRMA

For complete information on the following classes, please see the Stanford Bulletin for the current academic year. Most courses at CCRMA also have their own websites (see `http://www-ccrma.stanford.edu/CCRMA/Courses/Courses.html`).

Courses offered at CCRMA include:

- **Music 15A. Topics in Interactive Computer-Music Performance.**

  For sophomores only. Real-time interactive performance for interested musicians combining composition, performance, MIDI instruments, and computer programming. Introduction to programming, composition of short pieces, moving beyond familiar styles. Prepares students for work in ensembles and CCRMA courses.

- **Music 120. Introduction to Composition and Programming using MIDI-Based Systems.**

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

- **Music 149. Instrumental Music with Electronics.**

  The link between "traditional" evaluation of instrumental, orchestral, and vocal music and the revolutionary world of the electronic studio occurs in works where the two are combined. The course focuses on such linking works, beginning with Stockhausen's contributions and moving on to the products of IRCAM (Boulez, Murail, etc) and elsewhere.

- **Music 151. Psychophysics and Cognitive Psychology for Musicians.**

  Basic concepts and experiments relevant to use of sound, especially synthesized, in music. Introduction to elementary concepts; no previous background assumed. Listening to sound examples important. Emphasis on salience and importance of various auditory phenomena in music.

- **Music 154. History of Electroacoustic Music.**

  Survey of works and techniques.

- **Music 192. Theory and Practice of Recording**
  - **192A. Foundations of Sound Recording Technology.**
    Topics: elementary electronics, physics of transduction and magnetic recording of sound, acoustic measurement techniques, operation and maintenance of recording equipment, recording engineering principles.
  - **192B. Advanced Sound Recording Technology.**
    Topics: digital audio including current media, formats, editing software, and post-processing techniques. Also, microphone selection and placement, grounding and shielding techniques, noise reduction systems and advanced multi-track techniques.
  - **192C. Session Recording.**
    Independent engineering of recording sessions.
- **Music 220. Computer-Generated Music**
  - **Music 220A. Fundamentals of Computer-Generated Sound.**
    Techniques for digital sound synthesis, effects, and reverberation. Topics: summary of digital synthesis techniques (additive, subtractive, nonlinear, modulation, wavetable, granular, spectral-modeling, and physical-modeling); digital effects algorithms (phasing, flanging, chorus, pitch-shifting, and vocoding); and techniques for digital reverberation.
  - **Music 220B. Compositional Algorithms, Psychoacoustics, and Spatial Processing.**
    Use of high-level programming as a compositional aid in creating musical structures. Studies in the physical correlates to auditory perception, and review of psychoacoustic literature. Simulation of a reverberant space and control of the position of sound within the space.
  - **220C. Seminar in Computer Music Research.**
    Individual projects in composition, psychoacoustics, or signal processing.
  - **220D. Research.**
    Independent research projects in composition, psychoacoustics, or signal processing.
- **Music 252. Seminar: Topics in Computer Music.**
  Various topics according to interest.
- **Music 253. Musical Information: An Introduction.**
  Explores the diverse kinds of the musical information used in sound, graphical, and analytical applications. Device-independent concepts and principles in music representation and musical research objectives (repertory analysis, performance analysis, theoretical models, similarity and stylistic simultaion) will be emphasized. Examples will be drawn primarily from Western art music.
- **Music 254. Musical Representation and Computer Analysis: Seminar.**
  Offers an opportunity for participants to explore issues introduced in Music 253 in greater depth and to take initiative for research projects related to a theoretical or methodological issue, a software project, or a significant analytical result.
- **Music 319. Research Seminar on Computational Models of Sound Perception.**
  CCRMA hosts a weekly Hearing Seminar. All areas related to perception are discussed, but the group emphasizes topics that will help us understand how the auditory system works. Speakers are drawn from the group and visitors to the Stanford area. Most attendees are graduate students, faculty, or local researchers interested in psychology, music, engineering, neurophysiology, and linguistics.

- **Music 320. Introduction to Digital Signal Processing (DSP) and the Discrete Fourier Transform (DFT).**

  Introduction to the mathematics of digital signal processing and spectrum analysis for music and audio research. Topics: complex numbers, sinusoids, spectra, aspects of audio perception, the DFT, and basic Fourier time-frequency relationships in the discrete-time case.

- **Music 420. Spectrum analysis and signal processing using the FFT, with emphasis on audio applications.**

  Topics: FFT windows; cyclic and acyclic convolution; zero padding and other spectrum analysis parameters; the overlap-add and filter-bank-summation methods for short-time Fourier analysis, modification, and resynthesis; tracking sinusoidal peaks across FFT frames; modeling time-varying spectra as sinusoids plus filtered noise; FFT-based sound synthesis; brief overviews of and introductions to transform coders, perfect-reconstruction filter banks, and wavelet transforms.

- **Music 421. Signal Processing Methods in Musical Acoustics.**

  Computational models of musical instruments primarily in the wind and string families based on physical models implemented using signal processing methods. The models are designed to capture only the "audible physics" musical instruments using computationally efficient algorithms. Topics: mass-spring systems and their discrete-time simulation, sampled traveling waves, lumping of losses and dispersion, delay-line interpolation methods, applications of allpass filters and lattice/ladder digital filters in acoustic models, models of winds and strings using delay lines, scattering junctions, digital filters, and nonlinear junctions implementing oscillation sources such as bow-string and reed-bore couplings.

- **Music 422. Perceptual Audio Coding.**

  The need for significant reduction in data rate for wide-band digital audio signal transmission and storage has led to the development of psychoacoustics-based data compression techniques. In this approach, the limitations of human hearing are exploited to remove inaudible components of audio signals. The degree of bit rate reduction achievable without sacrificing perceived quality using these methods greatly exceeds that possible using lossless techniques alone. Perceptual audio coders are currently used in many applications including Digital Radio and Television, Digital Sound on Film, and Multimedia/Internet Audio. In this course, the basic principles of perceptual audio coding will be reviewed. Current and future applications (e.g. AC-3, MPEG) will be presented. In-class demonstrations will allow students to hear the quality of state-of-the-art implementations at varying data rates and they will be required to program their own simple perceptual audio coder during the course.

- **Music 423. Digital Signal Processing Research Seminar.**

  Ongoing seminar for doctoral students pursuing research in DSP applied to music or audio.

4.2 Workshops

CCRMA also offers a series of two-week summer workshops open to participants outside the Stanford community. Information regarding courses to be offered during the coming summer can be accessed from the CCRMA WWW Home Page[2]. Courses offered during the last few summers have included the following:

- **Digital Signal Processing for Audio: Spectral and Physical Models**

  This course covers analysis and synthesis of musical signals based on spectral and physical models. It is organized into morning lectures covering theoretical aspects of the models, and afternoon labs. The morning lectures present topics such as Fourier theory, spectrum analysis, the phase

[2] http://www-ccrma.stanford.edu/

vocoder, digital waveguides, digital filter theory, pitch detection, linear predictive coding (LPC), and various other aspects of signal processing of interest in musical applications. The afternoon labs are hands-on sessions using SMS, the Synthesis Toolkit in C++, SynthBuilder, and other software systems and utilities. The lectures and labs are geared to a musical audience with basic experience in math and science. Most of the programs used in the workshop are available to take.

- **Audio and Haptic Components of Virtual Reality Design**

  This course will introduce concepts and apply tools from cognitive psychology to the composition of virtual audio and haptic environments. In particular, the salience of various auditory and haptic phenomena to the perception and performance of music will be examined.

  Just as visual artists spend time learning perspective to provoke 3D effects, composers and virtual object designers must study the perceptual sciences to create virtual environments which are convincing upon hearing and touch. We will study relevant topics from acoustics, psychology, physics and physiology. We will apply these to the design and rendering of virtual objects not for the eyes, but for the haptic and audio senses. Principles of speech, timbre, melody, pitch, texture, force, and motion perception will be addressed. Various audio and haptic effects and illusions will be demonstrated.

  Morning lectures will cover these topics and also feature talks by eminent researchers and entrepreneurs working in the fields of psychoacoustics and haptics. Afternoon labs will provide practical experience in psychophysics experiment design and execution. In addition to sound synthesis tools, various haptic interfaces will be made available for experiment designs.

- **Interactive Composition and Performance with Computers**

  This introductory course will explore new approaches to interaction and improvisation between composer, performer, and computer. Topics to be discussed include performance interaction strategies (techniques of synchronization, timing, cueing, and parametric control), interactive algorithms, simulating live performance situations, tempo tracking, pitch following, and performance modeling.

  Hands on participation will use the Max programming environment and Common Music, a language that runs on Macintosh, PC and Unix based platforms. It will also involve real-time interaction using the Mathews-Boie Radio Baton (MIDI conductor/controller device). This course is particularly geared towards performers with an interest in interactive performance, improvisation and other ventures into the world of music technology. Emphasis will be on group performance projects, composition of new works, and realizations of existing interactive works.

- **Introduction to Sound Synthesis and Signal Processing Using CLM**

  This is an introductory and fast-paced workshop in sound synthesis techniques and digital audio effects, and their implementation in the CLM (Common Lisp Music) environment. We design software instruments that implement additive synthesis, subtractive, FM, sampling, wavetables, granular, spectral and physical modeling synthesis; and digital effects algorithms such as phasing, flanging, chorus, distortion and reverberation. Introductory signal processing and perception topics will be included.

  Common Lisp Music (CLM) is a public domain sound design language written on top of Common Lisp, currently running in Macintosh PowerPCs and several UNIX environments including SGI, Sun, NeXT and PC's running Linux. The workshop includes a Common Lisp lab that will teach basic Lisp programming skills. Familiarity with computers and programming languages is helpful but programming proficiency is not required.

- **Introduction to Algorithmic Composition**

  This course introduces basic principles and techniques of algorithmic composition and covers such topics as object oriented music representation, chance composition, musical automata and pattern languages. Sound synthesis used in the course material will include MIDI and Common Lisp Music. The course will be taught using the Common Music environment on Mac and NeXT workstations.

The workshop will be divided into morning lectures and afternoon lab times. During the lab hours the students will gain a hands-on experience working through projects and examples first presented in the morning lecture. All source code and documents from the workshop are free to take. Participation in Introduction to Sound Synthesis workshop or familiarity with Lisp is necessary for taking the workshop. Other prior programming experience is useful but not required.

- **Advanced Projects in Algorithmic Composition**

  A continuation of the above course, emphasis is placed on developing programming skills while working on individual projects.

- **Computers in Music Scholarship (Music Theory, Analysis, History, Ethnomusicology)**

  This course provides a comprehensive introduction to computer-assisted music research using the Humdrum Toolkit. Participants will learn to manipulate computer-based scores, tablatures, and other documents in order to solve a wide variety of analytic problems. By way of example, participants will learn to characterize common patterns of orchestration in Beethoven symphonies, examine harmony and voice-leading in Bach chorales, and investigate text/melody relationships in Gregorian chant.

  Thousands of full scores will be available for processing on-line – including repertoires from various cultures, periods, and genres. The course will be of particular value to scholars contemplating graduate level or advanced music research projects. The seminar staff will provide individual advice on participants' own research projects.

  All software and documentation from the workshop (including a sizeable score database) are free to take. The software is available for UNIX, DOS, OS/2 and Windows-95 (some restrictions apply). Familiarity with the 'emacs' or 'vi' text editors is recommended; limited knowledge of UNIX is helpful.

- **Intensive Audio Digital Signal Processing**

  This weekend-length workshop is specifically designed for engineers or developers working with audio who are interested in deepening their background in digital audio theory. The workshop covers the use of the Fast Fourier Transform (FFT) in digital signal processing, focusing on practical spectrum analysis, sound synthesis with spectral models, and signal processing using the FFT.

5 Compositional Activities

5.1 Overview

Since the late 60's most of the work in composition at CCRMA has been done in a software environment which evolved from the Music V program originally developed at Bell Labs by Max Mathews and his research group. The hardware and software has improved over the decades following, and the names of things have changed. Ported to a PDP10, Music V became the Mus10 music compiler system and played scores composed in Leland Smith's SCORE language. The compiler was replaced in 1977 with dedicated synthesis hardware in the form of the Systems Concepts Digital Synthesizer (built by Peter Samson and known as the "Samson Box"). The Samson Box was capable of utilizing many types of synthesis techniques such as additive synthesis, frequency modulation, digital filtering and some analysis-based synthesis methods. The PLA language, written by Bill Schottstaedt, allowed composers to specify parametric data for the Samson Box as well as for other sound processing procedures on the PDP10 mainframe (and on its eventual replacement, a Foonly F4). On April 3, 1992, the Foonly and Samson Box were officially retired. CCRMA has transitioned to a network of workstations (NeXTs, Intel based PCs, and SGI's) running NEXTSTEP, Linux, and Irix operating systems. The functionality of PLA exists now in the form of Common Music (CM) and STELLA (written in Common Lisp by Rick Taube), a software package that can write scores by listing parameters and their values, or by creating algorithms which then automatically determine any number of the parameters' values. Common Music (CM) can write scores in several different syntaxes (currently CLM, CMN, Music Kit, MIDI, CSound and Paul Lansky's real-time mixing program, RT). The scores can then be rendered on workstations using any of the target synthesis programs. For example, CLM (Common Lisp Music, written by Bill Schottstaedt) is a widely used and fast software synthesis and signal processing package that can make use of multiple Motorola 56001 DSPs or direct synthesis on faster workstations (including real time). CCRMA has also become the maintainer and distributor of NeXT's Music Kit, a real-time toolkit for computers running NEXTSTEP that merges the MIDI and real-time synthesis paradigms and can also be the target of Common Music generated scores.

Continuity has been maintained over the entire era. For example, scores created on the PDP10 or Samson Box have been recomputed in the NEXTSTEP computing environment, taking advantage of its increased audio precision. To summarize all these names for CCRMA's composing environment, the synthesis instrument languages have been, in chronological order, **MUS10, SAMBOX, CLM/MusicKit** and the composing language succession has been **SCORE, PLA, Common Music/Stella**. Other computers and software are also used 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 II Digital Audio Processor and several Macintosh II computers has brought renewed interest in real-time control and computer-based "musique concrète." The programming environments being used for composition and developmental research for all these systems include **MAX, LeLisp, Smalltalk, Common Lisp, DMIX** (a flexible compositional environment, written by Dan Oppenheim), **Objective C** and **C**.

Since its beginning, works composed at CCRMA have been highlighted at music festivals, concerts and competitions around the world. Recently, compositions realized at CCRMA were performed at the International Computer Music Conference in Banff; at the Society for Electro-Acoustic Music (SEAMUS) in the U.S.; at the Bourges Festival of Electroacoustic Music in France; at ISCM World Music Days; at The Warsaw Autumn Festival in Poland; at the Computers and Music Conference in Mexico City; at the Primera Muestra Internacional de Musica Electroacustica in Puerto Rico; and in concerts in Cuba, Greece, Russia, Argentina, Brazil, Spain, West Germany, Sweden, Switzerland, Italy, Hungary, and Czechoslovakia. Compositions from CCRMA have also won major electroacoustic music prizes over the past few years, including the NEWCOMP contest in Massachusetts, the Irino Prize for Chamber Music in Japan, the Ars Electronica in Austria, and the Noroit Prize in France. Works composed at CCRMA have been recorded on compact disks by Mobile Fidelity, Wergo, Harmonia Mundi, Centaur, and Allegro. CCRMA is publishing with Wergo/Schott *Computer Music Currents*, a series of 14 CDs containing computer music by international composers. *Computer Music @ CCRMA*, volumes one and two, were recently released. These two volumes represent music production by twelve composers working at CCRMA during the period 1992 to 1996.

5.2 Composers and Works

Recent compositional works realized at CCRMA include the following:

Celso Aguiar

- *All blue, I write with a blue pencil, on a blue sky*

  All blue ... was composed in 1996 for four-channel tape. This title was drawn from the writings of Walter Smetak (composer, instrument-builder, cellist and writer) to whose memory the piece is dedicated. The piece is about sound transformation, as a metaphor to the transformation of consciousness. Metallic percussion sounds are ever-present, while original cello sounds are broken into their rawest components. The basic cuisine for the piece was set up from these spices, and the dish is to be served hot. The cello has its identity transformed: its defining harmonic series is turned inharmonic, sounding closer to the metallic percussion. The pitches from this now bent, inharmonic series, are used as framework for a melodic-timbral game (the "blue pencil on a blue sky") played by cello and percussion. The cello transformations were obtained with SMSplus, a CLM system built on top of Xavier Serra's Spectral Modeling Synthesis and developed by the composer. A procedure for modeling the physical properties of a room via feedback-delay-networks was employed ("Ball within a Box", developed by Italian researcher Davide Rocchesso at CCRMA, with additional enhancements by the composer). *All blue...* won the 1997 "Premio SaoPaulo" at the 2nd International Electroacoustic Music Competition of Sao Paulo, Brazil and is available on volume one of the Computer Music @ CCRMA CD.

- *Monologue for Two*

  Monologue for Two for flute and clarinet (1993, revised and scored in 1997) is an investigation into unusual 'everyday life' facts. According to it, there is one day when you can't recognize an intimate friend, or you may suddenly realize you've become intimate to a most hostile enemy. Inasmuch, a dialogue can turn into monologue while still involving two players. The basic pitch materials in *Monologue for Two* were generated by computer programs in Daniel Oppenheim's Dmix environment for composition. The piece is dedicated to the memory of composer Ernst Widmer, who was quite aware of those everyday life 'compositional' facts. The piece is part of a cycle which also includes *Dialogue in One*, for piano. *Monologue for Two* received its first performance on Nov. 13th at the CCRMA 1997 Fall Concert in Campbell Recital Hall, Stanford University. The piece was performed by Karen Bergquist (flute), and Larry London (clarinet).

  Born in Palo Alto, California, Celso Aguiar grew up in Brazil in the town of Salvador, Bahia, where he studied composition with Swiss-Brazilian composer, Ernst Widmer. Since then he became interested in electronic music and went on to develop a computer-controlled digital synthesizer in Brazil. He is currently a DMA candidate in Composition at the Center for Computer Research in Music and Acoustics where he has been developing software tools for composition with spectral modeling, granular synthesis and sound spatialization. Celso Aguiar has written music for traditional instrumental as well as electronic media. His contact with composer Jonathan Harvey at Stanford has awakened in him a clear awareness for the spectral domain in music. Along with the skill for applying new DSP techniques, his compositional metier has been evolving towards an interesting amalgam of natural sounds and their most pungent transformations. His compositions have been performed in the Americas, Europe and Asia. His awards include both 1995 and 1997 "Premio Sao Paulo" at the 1st and 2nd International Electroacoustic Music Competition of Sao Paulo, Brazil, and a 1998 ICMA Commission Award.

Ron Alford

- *Girltalk*

  Girltalk is about my infatuation with children; what they think, and how they perceive our modern world. Children provide a fascinatingly uninhibited view, quite outside my adult reference, so I am forced to see things in a new light. This is the first of a series of related compositions using material from the world of children. This music was created algorithmically, using CM and CLM in a Linux environment (though the sound-sculpting was done on a Macintosh) during the summer of 1997.

  Ron Alford studied at the University of Illinois, the University of Colorado, Adams State College and at Stanford University. He has studied with George Crumb, Larry Hart, Wayne Scott, Vladimir Ussachevsky, and Cecil Effinger. He taught music in the American Southwest. He has been an active musician performing in symphony, chamber, jazz, church, rock, performance-art all his life. He has written, arranged, conducted, and judged music events. He has operated recording studios, hosted opera and 20th century music on commercial and NPR FM radio. He was founder of the New Mexico Jazz Workshop. He has been recipient of grants and National Endowment awards, and recently received an Arts Council award fellowship for Santa Clara County. His music has been heard in Canada, Austria, England, Denmark and Germany.

Jonathan Berger

- *The Voice Within the Hammer*
  - Version 1 - for flute, clarinet, mallet percussion, contrabass, piano and computer. Premiered, Mexico City, February 1997.
  - Version 2 - for flute, clarinet, viola, violin, piano and computer. Performances in Thessaloniki, Paris, New London, and Stanford.
- *Concerto for Piano and Orchestra*

  Jonathan Berger is a National Endowment for the Arts Fellow, 1996-1997.

Joanne D. Carey

- *Adventures on a Theme* (1997)
  1. The Unraveling
  2. Topsy Turvey and Haywire
  3. Harmony Rains

Adventures on a Theme is in three movements for flute and Radio-baton. This piece was premiered October 10th 1997 in San Diego at the Fourth Annual International New Music Festival, sponsered by the University of San Diego. The middle movement, called "Topsey Turvey and Haywire", is a radio-baton improvisation using programs that the composer developed with the help of Max Mathews last year. These programs define the response to the movements of the batons in the three dimensional space above the receiver box. The approach I have taken is to explore ways of varying a pre-composed melody in real-time with the batons. Ideally, the radio-baton and flutist would improvise together.

Joanne D. Carey is a visiting composer at CCRMA.

Chris Chafe

- *Push Pull* (1995)

  For celletto and live electronics. Performed in Buenos Aires; Hong Kong; Stanford; San Diego; Los Angeles. The celletto is the cellist's answer to all the fun keyboard players have been having lately with live computer synthesis. *Push Pull* is a setting for an "augmented player" where the computer almost becomes a part of the performer. Instrumental gestures are amplified musically and launch off into a life of their own. The soloist sows some of the seeds of what happens and can enter into dialogue with the musical textures that evolve. The work is a study for a new piece for singer and computer sponsored by a grant from the National Endowment for the Arts.

  The software system for PushPull has been applied in two small ensemble settings, both partly improvisational:

- *Triage*

  With Scott Walton for celletto, feedback guitar physical model, disklavier and computers. Premiered U.C. San Diego, 3 April 1997.

- *Between the Sheets*

  With Fred Malouf for electric guitar, tenor saxophone, celletto and computers. Premiered ICMC Thessaloniki, Greece, 27 Sep 1997.

  Chris Chafe is an awardee of National Endowment for the Arts Composer's Fellowship 1994-95; Green Faculty Fellowship 1995-96.

Kui Dong

- *YOULAN: Long Winding Valley* (1997)

  Youlan, for tape (two or more channels) and syncronized slides (done by visual artist Ruth Ecland), was realized at CCRMA between March 1-22, 1997. Youlan, a winding journey of exploration, is a term derived from classic Chinese poetry and music. The word connotes elements of the excitement of discovery, the lure of the unknown, and the elevation of the ordinary to a place of peak experience. The music is the map through this world, providing both context and direction. Samplings of ancient Chinese instruments have been transformed through digital processing and manipulation to create new sound structures that are evocative of their origins. The dynamic range of this piece is widely distributed: beginning with a highly tense drama, the piece slowly quiets down to an spiritually tranquil end after a series of sound material developments. The samples of steel plate Chinese instruments were processed and mixed using Spectral Modeling Synthesis (Xavier Serra and Raman Loureiro), Common Lisp Music (Bill Schottstaedt), and Real Time Mixer (Paul Lansky and K. Dickey) on a NeXT workstation at CCRMA, Stanford.

  Kui Dong was a winner of the 1994 Alea III International Composition Prize, Boston; 1990 National Art Song Competition; 1989 National Music and Dance Competition, Beijing; 1995 ASCAP Grant for Young Composers; 1995 Santa Clara Commissioning Award for Art; 1995 Djerassi Foundation for Art; 1993 Asia-Pacific National Fund; 1997 meet the composer/USA commissioning prgram award. Kui Dong is currently on the music faculty of Dartmouth University.

Michael Edwards and Marco Trevisani

- *segmentation fault beta 1.0* (1996)

  segmentation fault beta 1.0 is a composition for prepared and digitally processed piano, and computer mixed sound files. It uses software (artimix) written by Michael Edwards to trigger and mix sound files stored on hard disk. With this software, sounds are mapped to the keys of the computer keyboard and triggered at will during the performance. Each sound can also be mapped to a specific MIDI channel so that individual gain control can be applied to each sound in the mix through the use of a MIDI fader box. The computer part therefore consists of triggering prepared sounds and controlling their relative amplitudes. This piece is a collaboration between the two performers (Marco Trevisani, prepared piano, Michael Edwards, computer), both of whom are composers. The sounds used were created by the composers using Common Lisp Music, written by Bill Schottstaedt at Stanford University. They were realised with sample processing and manipulation of sounds from various sources, including piano, prepared piano and cello, as well as through direct synthesis using Frequency Modulation techniques. The piece was "upgraded" at the end of 1996 to *segmentation fault beta 1.1* and was performed at the Opus 415, No. 2 music festival in San Francisco. A multi-track studio recording was made in the summer of 1997.

R.J. Fleck

- *Essential Epiphanies* (1995)

  Part of an interactive performance environment employing movement, sound and sculptural forms, performed at Stanford's Memorial Auditorium and created at CCRMA; the result of a grant received from a consortium of Stanford arts faculty. Featuring a reading by vocalist and CCRMA-associate Emily Bezar, the sound design focused on the creation of soundscapes through the computer processing of previous readings of a composed text, and the real-time processing of both readings of the same text in performance, and other aspects of the performance environment. An early version of SynthBuilder was an essential element of the final performance configuration.

  Now living in San Francisco, R.J. is working to complete his program for the Doctorate of Musical Arts degree this year at CCRMA.

Nicky Hind

- *Cosmos* (1997)

  Cosmos is a composition for electronic synthesizers, radio baton, and computer. Designed for live, solo performance, the radio baton and computer keyboard are used as controllers in conjunction with custom-built software running on an Apple Power Macintosh computer. Using analog, frequency modulation, and sample playback synthesizers, the composition achieves gradual yet dramatic transformations of timbre and intensity which are activated by movements with the radio baton. Cosmos received its premiere at the CCRMA Summer concert in July 1997.

  Nicky Hind is presently working on further compositions to complement *Cosmos*, for an onging series of solo performances which will take place in a variety of venues and cultural contexts...

David A. Jaffe

- *Homage to Carl Sagan*

  A concerto for computer-processed Zeta violin and symphonic band.

- *The Seven Wonders of the Ancient World*

  A seventy-minute concerto in seven ten-minute movements for Boie-Mathews Radio Drum-controlled Disklavier Grand piano and ensemble of plucked string and percussion instruments: mandolin, guitar, harp, harpsichord, bass, 2 percussionists, harmonium. The piece will recieve its world premier by the San Francisco Contemporary Music Players on January 20th, 1998 at the Yerba Buena Theatre in San Francisco. Movements 1-4 were premiered with Sonor ensemble, UC San Diego, June, 1994; Movements 5-6 at the University of Victoria, April 1995. The fourth movement was also performed with Athelas ensemble, at ICMC 1994, Aarhus, Denmark. Work on the piece was supported in part by a Collaborative Composer Fellowship from the National Endowment for the Arts and the Banff Centre for the Arts, Canada. *The Seven Wonders of the Ancient World* was released on CD in 1996 on the Well-Tempered Productions label and was given an A+ rating by the Audio magazine.

Two statues, a temple, a roof-top garden, two tombs and a lighthouse. This rather odd collection of monuments has become famous as "The Seven Wonders of the Ancient World." All but one, the Pyramids, has been destroyed, either by Nature or by human hands. A closer look at the "Wonders" reveals a crosshatch of parallels and oppositions. Two deal with death-the Pyramids and the Mausoleum. The Hanging Gardens glorify cultivated nature, while Artemis was the goddess of wilderness and wild animals. The two statues are of the heavens - Zeus, the god of thunder and rain; and the Sun god of the Colossus of Rhodes.

How can the essence of these monuments be conveyed in music? In searching for an answer, the composer discovered two revolutionary instruments: the Yamaha Disklavier and the Mathews/Boie Radio Drum. The Disklavier is a modern version of the old player piano in that it can "play itself," while the Radio Drum is a percussion-like device that translates a percussionist's three-dimensional gestures into computer information. In 1992, the author conducted a series of experiments combining the Radio Drum and Disklavier and discovered that the flexible and seemingly magical mapping of percussion gestures onto piano sound makes possible the grand, monumental, yet very uncharacteristically "pianistic" sounds that had been looked for. The sound of this Drum-Piano is further expanded by an unusual orchestra consisting of instruments that extend the sound of the piano: harp, harpsichord, mandolin, guitar, bass, 2 percussionists, harmonium. Finally, an improvisational approach to the Drum Piano part allows the performer to respond and react to his unusual instrument. The result is a new kind of piano concerto.

David A. Jaffe received a National Endowment for the Arts Collaborative Fellowship in 1993-1994, an NEA Composer Fellowship in 1989 and an NEA Composer Fellowship in 1982. He was the NEA Composer-In-Residence with Chanticleer 1991. His music was recently featured on the San Francisco Symphony Chamber Music series at Davies Hall, where his string quartet "Quiet Places" was premiered.

Jun Kim

- *ZephyrBells* (1996)

  ZephyrBells is a composition for quadraphonic sound created using CLM (Common Lisp Music), SoundWorks and rt.app on the NeXT computer. I only used one sound source which is a synthetic bell sound for this piece. The basic idea is that we can hear the bell sounds from afar by the zephyr winds.

- *Dreaming* (1997)

  Dreaming is written for solo viola and computer generated tape sound. Its single movement consists of three sections. The first section can be described as Dreaming to Actuality; the second Actuality (viola solo); and finally a return to Dreaming. The source of tape sounds is entirely from an acoustic viola played by Keith Chapin, for whom the piece is written. The sound was processed using CLM (Common Lisp Music) on a NeXT computer.

Peer Landa

- *Gag Order*

  This piece was commissioned by NoTAM for the GRM Acousmonium sound system. The material is derived solely from three old native Japanese instruments: flute, metal-clock, and drum – and then rigidly processed by custom made DSP-applications I wrote exclusively for the piece.

Fernando Lopez Lezcano

- *House of Mirrors*

  "...come, travel with me through the House of Mirrors, the one outside me and the one within. Run, fly, never stop ... never think about being lost in the maze of illusions, or you will be. Glide with me through rooms, doors and corridors, surfing on tides of time, looking for that universe left

behind an eternity ago. Listen to the distorted steps, the shimmering vibrations that reflect in the darkness, watch out for the rooms with clocks where time withers and stops ..." fll.

House of Mirrors is an improvisational tour through a musical form and a four channel sound environment created by the composer/performer Fernando Lopez-Lezcano. The sound of doors opening and closing define the transitions between rooms, corridors and open spaces, where soundfile playback and midi controlled synthesis mix to create different atmospheres sharing a common thread of pitches, intensities and timbres. The journey through the House of Mirrors is controlled in real time through an interactive improvisation software package - PadMaster - developed by the composer over the past three years. The Mathews/Boie Radio Drum is the three dimensional controller that conveys the performer's gestures to PadMaster. The surface of the Radio Drum is split by PadMaster into virtual pads, each one individually programmable to react to baton hits and gestures, each one a small part of the musical puzzle that unravels through the performance. Hits can play soundfiles, notes, phrases or can create or destroy musical performers. Each active pad is always "listening" to the position of the batons in 3D space and translating the movements (if programmed to do so) into MIDI continuous control messages that are merged with the stream of notes being played. The virtual pads are arranged in sets or scenes that represent sections of the piece. As it unfolds, the behavior of the surface is constantly redefined by the performer as he moves through the predefined scenes. The performance of *House of Mirrors* oscillates between the rigid world of determinism as represented by the scores or soundfiles contained in each pad, and the freedom of improvisation the performer/composer has in arranging those tiles of music in time and space.

Jonathan Norton

- *Snapshots on a Circle* (1997)

  For alto saxophone, cello, percussion and tape. While traveling - and wanting to remember the experience - photographs are usually taken of people and places. Not being the most diligent about getting photos developed, several trips usually get mixed together. Snapshots on a Circle is an aural collage of the moods and interactions of the people and places where they occurred in the photographs.

  The title, Snapshots on a Circle, has a double meaning . The first is more literal in the sense that several of the photographs were taken during an extended lunch at a cafe on a plaza. The second is more universal in that most travels, no matter how long or how far, eventually wind their way back to their point of origin.

  The tape portion of this piece was realized by sampling everyday environmental sounds and then processing them in CLM and SoundDesigner II on an Apple PowerPC. They were then compiled on the Dyaxis II using MultiMix 2.3.

- *Textures of a Question* (1997)

  For tape. What is a question? How are questions formulated? As the mind wrestles to grasp the concept of a subject, inevitably, questions begin to form . But not all questions are created equal. Some may be ill-conceived and make no sense, resulting in more confusion. Some well thought out questions, once asked, can be enlightening but raise yet further questions. Some are really not questions at all, but simply a reiteration of the subject in the inquirer's own words in an attempt to understand. Sometimes frustration ensues, and the inquiry must be reapproached. From thought to vocalization this piece explores the musical texture of a question.

  This piece was realized through the use of granular synthesis, spectral reshaping and the resampling of vocal samples and computerized instruments. All signal processing was done on a Apple PowerPC.

Juan Carlos Pampin

- *Metal Hurlant* (1996)

  Metal Hurlant has been composed for a percussion player (playing metallic instruments) and computer generated sounds. The hybridity of the piece serves a qualitative logic. Atonal music during the '20s and serialism later stressed what Adorno referred to as the inner logic of procedures. In contrast, this work follows the logic of the sound materials, not the logic of the procedures, to shape acoustic matter. The acoustic material comes from a studio recording of metallic percussion instruments. Spectral analysis of these sounds provides the raw matter for the composition. This data is a digital representation of the qualitative traits of metallic percussion. It defines the range of acoustic properties available for manipulation and determines the further behavior of qualitative traits in the overall composition. In this way, qualitative parameters supply compositional parameters.

  Spectral analysis was used to explore what can be called the sound "metalness" of the selected instruments. Since the range of compositional operations is provided by the isolated sound metalness, to certain extent the qualitative structure of the material takes command over the compositional process. Moreover, the metalness ruling the computer generated sounds furnishes the morphological boundaries of theinstrumental part. Metal Hurlant is an expression of metalness sculpted on percussion and electronic sounds.

  The electronic sounds for this piece were generated with Bill Schottstaedt's CLM using my ATS library for spectral analysis, transformation and resynthesis (see research activities).

- *Interstices* (1997)

  > "They know that a system
  > is nothing more than
  > the subordination
  > of all aspects of the universe
  > to any one such aspect."
  > - from *Tlon, Uqbar, Orbis Tertius* by Jorge Luis Borges

  Sound. Sound as a metaphor of life, as a living entity that gets transformed with us, inside us, in our memory. *Interstices* is a journey inside sound, an aesthetic exploration of its components. In *Interstices* the string quartet meets electronic music in a poetic landscape, instruments become sometimes filters, sometimes synthesizers, not just imitating superficially these electronic devices, but abstracting their functionality: giving form to sound, operating on matter. Musical morphologies in *Interstices* may be seen as the reflections of the interior of a complex sound object, expressed in different time spans: short transients are stretched becoming long unstable sequences, instruments modulate stable portions of the sound illuminating regions of its spectrum. These sound-paths take the form of processes, they evolve in different layers and invite to be listened in many ways. Take the path you prefer, and enjoy your trip.

- *Toco Madera* (1997) for wooden percussion (duo) and computer generated sounds

  North of San Francisco, near Point Arena, the sea transforms the beach into a beautiful, constantly evolving mile long sculpture. On the beach hundreds of wood logs are washed onto the coast by the Pacific Ocean. I discovered this sculpture (or is it an installation?) while beginning work on *Toco Madera*. The dense textures created by drift wood of all sizes inspired the form and process of the piece. I realized that my compositional work had to be similar to the role of the sea, which not only placed the objects in textural combinations, but transformed their surfaces and matter to create new complex morphologies.

  I sculpted new sounds with the computer from a set of nine wooden percussion instruments recorded in the studio. I wanted to keep the rustic quality of wood sounds, to operate on them respecting their soul. This task was achieved using spectral analysis of the instrumental sounds to extrapolate their salient acoustic qualities, and digital filters to carve their matter. Throughout the piece, these transfigured wood sounds interact with the original instrumental set, performed by two percussion players, to create a multilayered musical space that reflects the textural traits of the natural wooden sculpture.

Toco Madera is the second of a cycle of percussion works exploring what philosopher Valentin Ferdinan calls "materiality" of sound. For this work (as for *Metal Hurlant*, the first piece of this cycle) a qualitative logic that guided the compositional process was inferred from the acoustic structure of the material used. In *Toco Madera* music becomes the expression of wood.

The analysis and spectral transformations of the instruments were done using ATS. All the digital signal processing for the piece was performed with Bill Schottstaedt's Common Lisp Music.

Born in Buenos Aires, Argentina, in 1967, Juan Pampin holds a Master in Computer Music from the Conservatoire Nationale Superieur de Musique de Lyon (SONVS). As a Visiting Composer at CCRMA in 1994, he composed the tape piece "Apocalypse was postponed due to lack of interest" that received an award in the Concours International de Musique Electroacoustique de Bourges 1995. He has been composer in residence at the LIEM-CDMC studio in Madrid. His music, including works for acoustic instruments, tape and mixed media, has been performed in the United States, Latin America, Europe and Asia.

Craig Stuart Sapp

- *Responsoria prolixia* (1997)

  An algorithmic-composition inspired piece for percussion quartet. The title of the composition derives from *responsoria prolixia* (great responsories) which are a prominent feature of Matins in the Office. The performers play a large role in shaping the form of the piece. There is always one performer at any given time who leads the other three through the composition which consists of 20 phrases of between three to eight beats. The leader informs the other performers of which phrase will be played next by sounding a unique two-beat rhythmic pattern. Two beats before the end of the phrase, the leader again chooses a new phrase, etc. A leader can pass-off their leadership role to another performer by playing a specific pattern.

  There are five levels of phrases in the composition. The first phrase level contains 3 beats, and then each additional phrase level adds another beat. Each time a leader gains control of the composition, he/she may select phrases from the next higher level; for example, if a performer has just become leader for the second time, that performer may choose to play any of the phrases in levels 1 and 2. Once the original leader has become leader for the sixth time, that performer may choose to end the composition with one of three cadences.

Kotoka Suzuki

- *Eclipse* (1996)

  A collaboration work for computer generated tape, dance, and pre-programmed robotic lights. Performed at Little Theater, Stanford California: Roble Dance Studio, Stanford, California.

  Kotoka Suzuki studied composition with Samuel Dolin at the Royal Conservatory of Music in Toronto between 1984 and 1989. She received a B.M. degree in composition from Indiana University in 1994 and is now working towards a D.M.A. degree at Stanford where she studies with Jonathan Harvey and David Soley. She has also been a fellow composer at Domain de Forget, June in Buffalo, and Voix Nouvelles music festivals, where she studied with York Hller, Walter Zimmermann, Brian Ferneyhough, and Franco Donatoni. Her works have been performed by the Stanford String Quartet, Nouvel Ensemble Moderne (Canada), and members of the Buffalo Symphony Orchestra. She has composed for both acoustic and electronic means, as well as for dance. She is currently working on her final project which will be for two percussionists, computer generated tape, and two dancers with real-time control sensors.

Marek Zoffaj

- *Voice* (1997)

  This composition is based on the combination of samples of human breathing, as a basic sign of human life, with music sequences and live soprano sequences. There is a very simple time line, where each "physical" gesture of the real world - a breath - is a source or an anticipation for the musical event. This dialog, which can literally be described as a contact of the real and imaginary world along two planes of the human mind, was the initial idea that started the work on this piece.

  Voice was first realized in the Experimental Electroacoustic Studio of the Slovak Radio in Bratislava and performed in Bourges (1997) and Bratislava (CEM 1997). It is now "under reconstruction" at CCRMA.

  Marek Zoffaj is a visiting scholar at CCRMA, Stanford University, as an awardee of the Fulbright Foundation.

6 Research Activities

Computer music is a multi-disciplinary field. The research summarized in this overview spans such areas as engineering, physics, computer science, psychology, and music (including performance, analysis, and composition). Any given research topic may require sophistication in several of these fields. This document can only contain a brief review of the work being done at CCRMA. For a more complete description of the research, a list of CCRMA research publications is included. Copies of reports are available upon request.

The researchers working at CCRMA include graduate students, faculty, staff, and visiting scholars. Email may be sent to any of these people by mailing to ***login*@ccrma.stanford.edu** where *login* names are listed in the roster at the beginning of this publication.

6.1 Computer Music Hardware and Software

6.1.1 SMSPlus: Post-Processed Real-Time SMS Instruments in CLM

Celso Aguiar

This project is an adaptation of Xavier Serra's Spectral Modeling Synthesis (SMS) technique for compositional purposes. It provides an SMS sound composition environment integrating several tools: First, the sound is analyzed from a Unix shell using Serra's C programs. A graphical interface (SMSEditor, from an Objective C prototype by Serra) has been greatly enhanced in order to display the resulting files in a three-dimensional waterfall plot. After the analysis is done, several routines support reading and writing of SMS files from inside MatLab (cmex files) and the post-processing and normalization of these files. Once analysis and post-processing are done, a series of routines and instruments integrating Lisp, CLM (Bill Schottstaedt) and C, are used for the resynthesis of the sound. The resynthesis employs the Inverse FFT algorithm (Xavier Rodet) which Xavier Serra and I programmed in the '94 Summer Workshop at CCRMA. The resynthesis programs run in real-time.

6.1.2 The CCRMA Music Kit and DSP Tools Distribution

David Jaffe and Julius Smith

The 4.2 version of the Music Kit was released in 1997 and is available free of charge via FTP at `ftp://ccrma-ftp.stanford.edu/pub/NeXT/MusicKit/`. This release is compatible with NEXTSTEP software releases 3.2 and later on NeXT and Intel-based hardware. Also, Music Kit programs that are compiled under NEXTSTEP can run on OPENSTEP for Intel and NeXT hardware.

Release 4.2 is an incremental release with several significant additions:

- SUPPORT FOR TURTLE BEACH FIJI/PINNACLE DSP CARDS (INTEL-BASED HARDWARE)

  The 4.2 Music Kit includes a new driver and support for the Turtle Beach Fiji and Pinnacle DSP cards. These cards provide the best price/performance of any currently-available Music Kit-compatible DSP cards (as of July 1997). They have a DSP56002, 32K of fast static RAM, and both digital and high-quality analog I/O. The Pinnacle also has an MPU401-compatible Kurzweil synthesizer that will work with the Music Kit MIDI driver. In addition, the Music Kit driver for the Turtle Beach Multisound, Tahiti and Monterrey has been upgraded to support the full Turtle Beach DSP memory space.

- UPGRADED INTEL-BASED HARDWARE SUPPORT

  The Intel implementation has been optimized. Support for writing soundfiles from the DSP is now supported on Intel hardware. This functionality was previously available only on NeXT hardware.

- NEW APPLICATIONS

  Two Music Kit applications of note are available separately:

  - *Sequence*, a Music Kit Sequencer developed by Pinnacle Research. Available free from the CCRMA ftp server (`ftp://ccrma-ftp.stanford.edu/pub/NeXT/Sequence.9.8.4.tar.Z`). This was released to the Net in the summer of 1996. The ftp site now has an updated version that includes the Fiji/Pinnacle support.
  - *SynthBuilder*, a synthesis instrument design and performance tool. SynthBuilder was the Grand Prize winner of the Second Annual International Music Software Competition at Bourges. It was developed by Stanford University's Sondius program, and is now being supported and further developed by Staccato Systems Inc. The NEXTSTEP version, including a free license authorization code, is available from `http://www.StaccatoSys.com` or `ftp://ftp.StaccatoSys.com`. Staccato Systems is also porting SynthBuilder to Windows 95, using the host CPU to do synthesis.

- NEW UNIT GENERATORS

  There are a variety of new UnitGenerator classes. For example, rock-solid real-time envelopes are now available with AsympenvUG, which down-loads its envelope to the DSP, instead of feeding the break-points down one at a time (as does AsympUG.)

- OTHER NEW FEATURES

  For more details on these items, as well as other new features, please see the Music Kit release notes, which are available via ftp at `ftp://ccrma-ftp.stanford.edu/pub/NeXT/MusicKit/ReleaseNotes.rtf`.

Other Music Kit News

Until recently, we were making extensive use of the "Frankenstein" cards (in various forms), home-brewed DSP cards based on the Motorola EVMs. However, with the advent of the Turtle Beach Fiji and Pinnacle cards, we no longer feel it is necessary (or worth the trouble) to pursue the "Frankenstein" direction.

We have been planning to provide a combined sound/MIDI driver for SoundBlaster-compatible cards. We negotiated with NeXT to do this (because we needed permission to use their sound driver code) and everything was ready to happen, but then there were some legal complications that held things up, so we weren't able to get this done for the 4.2 release.

Plans are in the works to port at least some of the Music Kit to Rhapsody.

Music Kit Background

The Music Kit is an object-oriented software system for building music, sound, signal processing, and MIDI applications in the NEXTSTEP programming environment. It has been used in such diverse commercial applications as music sequencers, computer games, and document processors. Professors and students have used the Music Kit in a host of areas, including music performance, scientific experiments, computer-aided instruction, and physical modeling. The Music Kit is the first to unify the MIDI and Music V paradigms, thus combining interaction with generality. (Music V, written by Max Mathews and others at Bell Labs three decades ago, was the first widely available "computer music compiler.")

The NeXT Music Kit was first demonstrated at the 1988 NeXT product introduction and was bundled in NeXT software releases 1.0 and 2.0. Since the NEXTSTEP 3.0 release, the Music Kit has been distributed by CCRMA. Questions regarding the Music Kit can be sent to `musickit@ccrma.stanford.edu`.

The CCRMA Music Kit and DSP Tools Distribution (or "Music Kit" for short) is a comprehensive package that includes on-line documentation, programming examples, utilities, applications and sample score documents. It also comes with Bug56 (black hardware only), a full featured, window-oriented, symbolic debugger by Ariel Corp. for the Motorola DSP5600x signal processing chip family.

6.1.3 Common Music

Heinrich Taube

What is Common Music?

Common Music (CM) is an object-oriented music composition environment. It produces sound by transforming a high-level representation of musical structure into a variety of control protocols for sound synthesis and display: MIDI, Csound, Common Lisp Music, Music Kit, C Mix, C Music, M4C, RT, Mix and Common Music Notation. Common Music defines an extensive library of compositional tools and provides a public interface through which the composer may easily modify and extend the system. All ports of Common Music provide a text-based music composition editor called Stella. A graphical interface called Capella currently runs only on the Macintosh. See `http://www-ccrma.stanford.edu/CCRMA/-Software/cm/cm.html` for more information.

History

Common Music began in 1989 as a response to the proliferation of different audio hardware, software and computers that resulted from the introduction of low cost processors. As choices increased it became clear that composers would be well served by a system that defined a portable, powerful and consistent interface to the myriad sound rendering possibilities. Work on Common Music began in 1989 when the author was a guest composer at CCRMA, Stanford University. Most of the system as it exists today was implemented at the Institut fr Musik und Akustik at the Zentrum fr Kunst und Medientechnologie in Karlsruhe, Germany, where the author worked for five years. Common Music continues to evolve today at the University of Illinois at Urbana-Champaign, where the author is now a professor of music composition. In 1996 Common Music received First Prize in the computer-assisted composition category at the 1er Concours International de Logiciels Musicaux in Bourges, France.

Implementation

Common Music is implemented in Common Lisp and CLOS and runs on a variety of computers, including NeXT, Macintosh, SGI, SUN, and i386. Source code and binary images are freely available at several internet sites. In order to compile the source code you need Common Lisp. The best implementations are commercial products but there are also several good public domain implementations available on the Internet. See `http://www-ccrma.stanford.edu/CCRMA/Software/cm/cm.html` for more information.

Synthesis Control

Each synthesis target is represented as a "syntax" in Common Music. Any combination of syntaxes can be included when the system is built from its sources. The available syntaxes are:

Synthesis Target	Syntax	Works on
C Mix	CMIX	everywhere
C Music	CMUSIC	everywhere
C Sound	CSOUND	everywhere
Common Lisp Music	CLM	NeXTstep, Linux, IRIX
Common Music Notation	CMN	everywhere
M4C	M4C	NeXTstep
Mix	SGIMIX	IRIX
MIDI	MIDI	everywhere
Music Kit	MK	NeXTstep
RT	RT	NeXTstep, IRIX

Whenever possible, CM sends and receives directly to and from the target. Otherwise, a file can be generated and sent to the target automatically so that the process of producing sound appears seamless and transparent.

All ports of CM support reading level 0 and 1 MIDI files and writing level 0 files. Direct-to-driver MIDI input and output is supported for the following configurations:

Mac OS 7.x	MCL 2.0.1, 3.0
NeXTstep 3.2	ACL 3.2.1, 4.1; GCL 21.1; CLISP
Windows 3.1	ACL/PC

Contact

To receive email information about software releases or to track developments in CCRMA's family of Lisp music programs: CM, CLM and CMN please join `cmdist@ccrma.stanford.edu` by sending your request to `cmdist-request@ccrma.stanford.edu`.

6.1.4 SEE–A Structured Event Editor: Visualizing Compositional Data in Common Music

Tobias Kunze and **Heinrich Taube**

Highly structured music composition systems such as Common Music raise the need for data visualization tools which are general and flexible enough to adapt seamlessly to the—at times very unique—criteria composers employ when working with musical data. These criteria typically involve multiple levels of data abstraction and interpretation. A "passing note", for instance, is a fairly complex, *compound* musical predicate, which is based on properties of several other, lower–level musical predicates such as the degree of consonance, metric position, or melodic direction, all of which are of different complexity, draw upon different primitives, and apply only to a limited set of data types, that is, "notes". Visualizing compound musical predicates then translates to a mapping of a set of criteria—predicates and properties—on a set of display parameters.

The SEE visualization tool provides graphical and programming interfaces for these two tasks and consists of an abstracting program layer to allow for the construction of custom musical predicates out of a possibly heterogenous set of data and a separate program module which controls their mapping onto a wide variety of display parameters. As large screens and full color support become more and more standard for most computer systems as well as to account for the complexity that comes with visualizing musical predicates in general, the display parameters make consequent use of both, color and the 3D visualization paradigm. Thus, object *position* as well as *extension* along the x, y, and z axes, object *representation* (model), and *color* (position of its color along the coordinate axes of the current color model) may be assigned up to ten or more predicates.

Although SEE may be used as a standalone tool, it is highly integrated and primarily intended to be used with Capella, Common Music's graphical user interface. The application framework itself and the programming interfaces are implemented in Common Lisp, and thus run on a variety of platforms.

The current version is being developed on a SGI workstation using the X11 windowing system and the OpenGL and OpenInventor graphics standards, but portability is highly desired and upcoming ports will most probably start out with the Apple Macintosh platform.

6.1.5 PadMaster, an Interactive Performance Environment. Algorithms and Alternative Controllers

Fernando Lopez Lezcano

PadMaster is a a real-time performance / improvisation environment currently running under the NextStep operating system. The system primarily uses the Mathews/Boie Radio Drum as a three dimensional controller for interaction with the performer, although that is no longer the only option. The Radio Drum communicates with the computer through MIDI and sends x-y position and velocity information when either of the batons hits the surface of the drum. The Drum is also polled by the computer to determine the absolute position of the batons. This information is used to split the surface of the drum into up to 30 virtual pads of variable size, each one independently programmable to react in a specific way to a hit and to the position information stream of one or more axes of control. Pads can be grouped into Scenes and the screen of the computer displays the virtual surface and gives visual feedback to the performer. Performance Pads can control MIDI sequences, playback of soundfiles, algorithms and real time DSP synthesis. The velocity of the hits and the position information can be mapped to different parameters through transfer functions. Control Pads are used to trigger actions that globally affect the performance.

The architecture of the system has been opened and it is now possible to create interfaces to other MIDI controllers such as keyboards, pedals, percussion controllers, the Lightning controller and so on. More than one interface controller can be active at the same time listening to one or more MIDI streams and each one can map gestures to the triggering and control of virtual pads. The problem of how to map different simultaneous controllers to the same visible surface has not been completely resolved at the time of this writing (having just one controller makes it easy to get simple visual feedback of the result of the gestures, something that is essential in controlling an improvisation environment). Another interface that is being currently developed does not depend on MIDI and controls the system through a standard computer graphics tablet. The surface of the tablet behaves in virtually the same way as the surface of the Radio Drum, and tablets that have pressure sensitivity open the way to three dimensional continuous control similar to that of the Radio Drum (but of course not as flexible). The advantage of this interface is the fact that it does not use MIDI bandwidth and it relies on hardware that is standard and easy to get.

Performance Pads will have a new category: Algorithmic Pads. These pads can store algorithms that can be triggered and controlled by gestures of the performer. While a graphical programming interface has not yet been developed at the time of this writing, the composer can create algorithms easily by programming them in Objective C within the constraints of a built in set of classes and objects that should be enough for most musical purposes. Any parameter of an algorithm can be linked through a transfer function to the movement of one of the axes of control. Multiple algorithms can be active at the same time and can respond in different ways to the same control information making it easy to transform simple gestures into complicated musical responses. An algorithm can also be the source of control information that can be used by other algorithms to affect their behavior.

6.1.6 A Dynamic Spatial Sound Movement Toolkit

Fernando Lopez Lezcano

This brief overview describes a dynamic sound movement toolkit implemented within the context of the CLM software synthesis and signal processing package.

The goal of the project is to provide a set of tools that can be used by a composer working within the CLM environment to represent, control and render multiple moving sound sources in, for now, bidimensional space. The user interface consists of a unit generator for controlling the spatial movement and the environment in which it occurs (dlocsig), and a set of CLOS classes for the description in simple terms of the path each sound object follows (path). The composer describes the movement by specifying a small set of points in space and the software routines generate interpolated Bezier curves that create a smooth trajectory between the specified points.

The low level hardware support for four channel reproduction is an external high quality four channel digital to analog converter box (the QuadBox) connected to the DSP port of a NeXT workstation. The corresponding software support consists of a C/DSP56000 assembler program that can play standard four channel soundfiles through the QuadBox.

Part of the work in this project was done as a team effort with Atau Tanaka at CCRMA while Fernando was working in Japan at the SFC Campus, Keio University.

6.1.7 ATS: Analysis/Transformation/Synthesis: A Lisp Environment for Spectral Modeling

Juan Pampin

ATS is a library of Lisp functions that perform spectral Analysis, several types of spectral Transformations, and Synthesis of sounds. As a Lisp environment for sound design, ATS allows the user to explore the possibilities of spectral modeling in a very flexible way by means of dynamic control of sound parameters. The synthesis engine of ATS is implemented using Common Lisp Music. Two re-synthesis algorithms are available, one performing additive synthesis using oscillators, and another implementing a Filter/Source (Subtractive) model, using an overlap add Inverse FFT technique (fft-filter). For more information about ATS please point your web browser to `http://www-ccrma.stanford.edu/~juan/ATS.html`.

6.1.8 *Ashes Dance Back*, a collaborative work with Jonathan Harvey

Juan Pampin

I collaborated with professor Jonathan Harvey for the sound design of his piece *Ashes Dance Back*, for choir and electronic sounds. This collaboration was four quarters long, covering fall/winter 95-96, and fall/winter 96-97. At the request of professor Harvey I used my ATS system (see "Current Research Activites" section) for spectral modeling to generate the electronic sounds of the piece based on the analysis and transformation of a single vocal sound: a B flat sample of my own singing.

During the composition of this piece, many improvements and additions were done to ATS. Here is a list of the most prominent ones:

- A cross synthesis algorithm was implemented based on a subtractive synthesis model, allowing the generation of hybrid materials by crossing natural sounds (wind, fire, water, etc.) with vocal sounds.
- A missing fundamental search algorithm was developed to match arbitrary clusters of partials to a virtual fundamental frequency, taking account of masking and loudness features of the partials.
- A spectral shifting operator was created allowing the generation of new spectra based on frequency shifts of an original vocal spectrum sample. A set of nine spectral compressions was generated by this mean. These compressions served as the harmonic structure for the whole piece.

The equalization, montage, and final mix of all the electronic materials was done using CLM. For the performance of the electronic sounds of the piece we used the following strategy: long sequences (most of them backgrounds) were stored on CD and triggered by the sound engineer in the concert. Medium to short materials (1 to 20 second long) where transferred to two Emu E64 samplers and interpreted by a keyboard player during performance. Ashes Dance Back was premiered at the Strasbourg Musica Festival on September 27, 1997.

6.1.9 Stanford Computer-Music Packages for Mathematica

Craig Stuart Sapp

The Webpage `http://www-ccrma.stanford.edu/CCRMA/Software/SCMP` contains links to various Mathematica packages dealing with computer music topics. The main package, SCMTheory, contains visualization and manipulation tools dealing with the fundamentals of digital signal processing, such as complex numbers, plotting complex domains and ranges, and modulo sequences and manipulations. The Windows package contains the definitions of various analysis windows used in short-time fourier transform analysis. The FMPlot package contains functions for plotting simple FM-synthesis spectra.

All packages run with Mathematica version 2.0 or greater, except the Windows package which requires Mathematica 3.0. Included on the SCMP main webpage are Mathematica notebooks which demonstrate various aspects of the SCMP set of packages. Also included on the SCMP main webpage are these notebooks in PDF format for viewing by those people who do not have Mathematica.

6.1.10 Graphical Additive Synthesis

Craig Stuart Sapp

A command-line program, `line2sine`, was written to interpret graphic lines in a CAD-like drawing program as sinwaves. Documents created by the NEXTSTEP program Diagram.app are read by the `line2sine` program and any lines in that document are converted into freqency and amplitude envelopes which are then fed into oscillator unit-generators. The `line2sine` program can be downloaded from `ftp://ftp.next.peak.org/next/apps/soundapps/line2sine.1.0.NI.bs.tar.gz` or `ftp://peanuts-.leo.org/pub/comp/platforms/next/Audio/programs/line2sine.1.0.NI.bs.tar.gz`. These two files contain the program, documentation, and examples. On-line documentation can be found at `http://hummer.stanford.edu/sig/examples/line2sine/doc`, and on-line sound/graphics examples can be found at `http://hummer.stanford.edu/sig/examples/line2sine/doc/examples`.

6.1.11 Common Lisp Music, Snd and Common Music Notation

William Schottstaedt

Common Lisp Music (CLM) is a sound synthesis package in the Music V family written primarily in Common Lisp. The instrument design language is a subset of Lisp, extended with a large number of generators: oscil, env, table-lookup, and so on. The run-time portion of an instrument can be compiled into C or Lisp code. Since CLM instruments are lisp functions, a CLM note list is just a lisp expression that happens to call those functions. Recent additions to CLM include support for real-time interactions and integration with the Snd sound editor.

The Snd sound editor is a recent addition to the Common Music package. It is modeled loosely after Emacs and an old, sorely-missed PDP-10 editor named Dpysnd. It can accommodate any number of sounds, each with any number of channels. Each channel is normally displayed in its own window, with its own cursor, edit history, and marks; each sound has a control panel to try out various changes quickly; there is an overall stack of 'regions' that can be browsed and edited; channels and sounds can be grouped together during editing; and edits can be undone and redone without restriction.

Common Music Notation (CMN) is a music notation package written in Common Lisp, using CLOS and the Sonata font. It provides for all the usual needs of music notation in a fully customizable, programmable environment.

CLM, CMN, and Snd are available free, via anonymous ftp at `ftp://ftp-ccrma.stanford.edu` as pub/Lisp/clm.tar.gz, pub/Lisp/cmn.tar.gz, and pub/Lisp/snd.tar.gz.

6.1.12 SynthBuilder, SynthScript, and SynthServer—Tools for Sound Synthesis and Signal Processing Development, Representation, and Real-Time Rendering

Julius Smith, David Jaffe, Nick Porcaro, Pat Scandalis, Scott Van Duyne, and Tim Stilson

The SynthBuilder, SynthScript, and SynthServer projects have been spun out from CCRMA to a new company Staccato Systems, Inc. The tools are currently being ported to "all major platforms" and focused into specific software products. Watch the Staccato website[3] for latest details.

[3] http://www.staccatoSys.com

6.1.13 Tactile Manipulation of Software

Sean Varah

Extending existing software at CCRMA to incorporate tactile manipulation of software. My work at the Harvard Computer Music Center involved adapting computer music software to emulate analog studio techniques. I plan to adapt digital signal processing programs to accept MIDI or other external controller information to change program parameters. For example, an on-screen digital filtering program would have its frequencies, bandwidth, and attenuation set by MIDI sliders, so a composer could manipulate parameters in a tactile way, emulating analog graphic equalizers. By setting up external controllers, the composer would then be able to manipulate several parameters at once, as opposed to typing single parameters, or adjusting one parameter at a time with the mouse. I then plan to use this type of interactive control in live performance.

6.2 Physical Modeling and Digital Signal Processing

6.2.1 Physical Modeling of Brasses

David Berners

One of the difficulties in building waveguide models of brasses and winds is that we do not know how to find the round-trip filtering in a flaring horn without actually making an acoustic measurement. Ideally, we would like to be able to compute the loop filter directly from the physical dimensions of the horn. While significant work has been done along these lines (Causse et al. [1], Plitnik and Strong [2], Benade [3]), a complete and accurate theory is not yet available.

To provide computationally tractable models, the flaring horn is modeled assuming that Webster's horn equation is satisfied, i.e., that a one-parameter solution to the wave equation exists within the boundaries of the horn. Any shape, such as planar or spherical, can be assumed for the wavefront within the horn.

In an ongoing research project at CCRMA, Webster's horn equation is solved as follows: First, the wave equation is converted to the form of the celebrated Schrodinger wave equation through a coordinate transformation outlined by Benade in [3]. Once in Schrodinger's form, the wave equation becomes equivalent to the one-dimensional scattering problem in particle physics, for which efficient and numerically stable solution methods exist (Kalotas and Lee [4]). In the new (transformed) coordinate system, the horn boundary function is replaced by the "horn potential function," which, in addition to providing the frequency dependent reflection, transmission, and impedance functions for the waveguide, can be used to gain an intuitive understanding of how these characteristics are related to bell flare. The quantities obtained from the solution to Webster's equation are all that is necessary for the design of lumped filters to be used in a digital waveguide model. Advantages over conventional modeling techniques include the ability to specify an arbitrary wavefront shape and possible numerical advantages.

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6.2.2 Synthesis of Transients in Classical Guitar Sounds

Cem Duruoz

Synthesis of acoustic musical instrument sounds using computers has been a fundamental problem in acoustics. It is well known that, the transients heard right before, during and right after the attack portion of an instrumental sound are the elements which give the instrument most of its individual character. Therefore, in a synthesis model, it is crucial to implement them carefully, in order to obtain sounds similar to those produced by acoustic instruments. The transients in classical guitar sounds were studied by making studio recordings, digital editing and Fourier Analysis. The sounds heard in the vicinity of the attack were classified according to the origin, spectral content and duration. Next, a hybrid FM/Physical Modeling Synthesis model was developed to produce these transients sequentially. The parameters such as the duration, amplitude and pitch were extracted from further recordings, and incorporated into the model to synthesize realistic classical guitar sounds.

6.2.3 Scalable Audio Models for Data Compression and Modifications

Scott Levine

The best methods currently for high quality, low bitrate audio compression algorithms are based on filterbanks. While current algorithms, such as MPEG-AAC (Advanced Audio Compression), achieve very high data efficiency, it is very difficult to perform modifications such as time stretching and pitch shifting on the compressed data.

In this study, we investigate a more flexible model for audio that allows competitive scalable data compression rates while allowing for simple modifications on the compressed data. Through a combination of multiresolution sinusoidal modeling, transient modeling, and noise modeling, we can achieve both a scalable, efficient audio data representation that is also easy to modify.

6.2.4 Articulatory Singing Voice Synthesis

Hui-Ling Lu

The goal of this research is to convert score files to synthesized singing voice. The framework is based on a library of control parameters for synthesizing basic phonemes, together with interpolation techniques for synthesizing natural phoneme transitions, tempo, and pitch.

The starting point for this work is the Singing Physical Articulatory Synthesis Model (SPASM), originally developed at CCRMA by Perry Cook. The SPASM software system is based on the "source-filter" paradigm: The glottal source (source part) is modeled by a parametric mathematical equation, and the vocal tract (filter part, which shapes the spectrum of the source) is simulated by a digital waveguide filter (DWF).

In this research, the interaction between the source and filter is extended by exploring more complicated glottal source models from the articulatory speech synthesis literature.

It turns out that the control parameter library construction is nontrivial. It includes the "inversion problem" which tries to retrieve the model parameters from the voice output signal only. The inversion problem is non-unique and nonlinear. Various existing methods from articulatory speech synthesis and some other general optimization methods are under evaluation.

6.2.5 Optimal Signal Processing for Acoustical Systems

Bill Putnam

Recent advances in optimization theory have made it feasible to solve a class very large scale optimization problems in an efficient manner. Specifically, if a problem can be shown to be /emphconvex, then one can make use of recent advances in interior point optimization methods to achieve optimal solutions to problems whose scale is beyond the capabilities of more traditional optimization techniques.

Many interesting problems in audio and acoustical signal processing can be shown to belong to the class of convex optimization problems. My research has focused on several of these problems.

- **Inverse Filtering of Room Acoustics – Echo Cancellation:** Previous research in the field, has shown that under certain conditions, one can achieve perfect cancellation of a rooms acoustic response using multiple sources. No technique has been presented to design a set of optimal filters to achieve this goal. My work in this area has been to apply convex optimization theory to determine an optimal set of filters.

- **Broadband Acoustical Arrays:** Beamforming using arrays of transducers operating at a specific frequency is a well understood and researched topic. Typical audio applications require an array to perform over a wide range of frequencies (typically 1-2 octaves). The problem becomes one of designing multiple filters (one for each transducer), to achieve the desired beam pattern over the frequency range of interest.

A real time system is being developed for both of the above applications. This system is capable of measurement, and subsequent implementation of a parallel bank of filters. The 'Frankenstein' hardware is used to allow for up to 16 separate channels of audio. A version of the software using commercially available DSP hardware will be available from `http://www.ccrma.stanford.edu/~putnam`.

- **Fractional Delay Filters:** In order to accurately *tune* a physical model of an musical instrument, delay lines with fractional sample delay are needed. To achieve this, one needs to implement a filter whose group delay is a fraction of a sample. Typical optimal filter design methods such as the Remez exchange do not extend to the case where the desired frequency is complex, which is necessary in this case. This problem is convex, and hence can be solved with interior point methods.

6.2.6 Sig++: Musical Signal Processing in the C++ language

Craig Stuart Sapp

Sig++ is a set of C++ classes for use to generate sound programs from flowgraphs of filters, etc. The paradigm for sig++ implementation is to imagine a flowgraph consisting of signal manipulators which are connected to each other to do something interesting. An intent of sig++ is to have very portable code. As a result, example programs using the sig++ library have been compiled on several computer configurations: NextStep, OpenStep, Linux, Sun SPARCStations, HP-UX, and SGI IRIX. A Visual C++ version may be developed soon.

See the main webpage for sig++ at `http://www-ccrma.stanford.edu/~craig/sig` which contains an overview, example binaries and sources, example sounds created by the example programs, documentation for the classes included in the sig++ library, as well as the source code for those classes.

Currently work is being done to incorporate real-time musical control into the sig++ libraries, primarily using the Linux operating system and 4Front Technologies' OSS soundcard drivers.

6.2.7 Acoustic Research and Synthesis Models of Woodwind Instruments

Gary P. Scavone

The modeling of musical instruments using digital waveguide techniques has proven to be both an accurate and efficient technique for synthesis. Because such models are based on physical descriptions, they further provide a useful tool for acoustical explorations and research.

Dissertation

Results of this work were recently published in *An Acoustic Analysis of Single-Reed Woodwind Instruments with an Emphasis on Design and Performance Issues and Digital Waveguide Modeling Techniques*, a Ph.D. thesis completed at CCRMA, Stanford University. In this study, current acoustic theory regarding single-reed woodwind instruments is reviewed and summarized, with special attention given to a complete analysis of conical air column issues. This theoretical acoustic foundation is combined with an empirical perspective gained through professional performance experience in a discussion of woodwind instrument design and performance issues. Early saxophone design specifications, as given by Adolphe Sax, are investigated to determine possible influences on instrument response and intonation. Issues regarding saxophone mouthpiece geometry are analyzed. Piecewise cylindrical and conical section approximations to narrow and wide mouthpiece chamber designs offer an acoustic basis to the largely subjective examinations of mouthpiece effects conducted in the past. The influence of vocal tract manipulations in the control and performance of woodwind instruments is investigated and compared with available theoretical analyses. Several extended performance techniques are discussed in terms of acoustic principles.

Discrete-time methods are presented for accurate time-domain implementation of single-reed woodwind instrument acoustic theory using digital waveguide techniques. Two methods for avoiding unstable digital waveguide scattering junction implementations, associated with taper rate discontinuities in conical air columns, are introduced. A digital waveguide woodwind tonehole model is presented which incorporates both shunt and series impedance parameters. Two-port and three-port scattering junction tonehole implementations are investigated and the results are compared with the acoustic literature. Several methods for modeling the single-reed excitation mechanism are discussed.

Expressive controls within the context of digital waveguide woodwind models are presented, as well as model extensions for the implementation of register holes and mouthpiece variations. Issues regarding the control and performance of real-time models are discussed. Techniques for verifying and calibrating the time-domain behavior of these models are investigated and a study is presented which seeks to identify an instrument's linear and nonlinear characteristics based on periodic prediction.

Current and Future Work

This area of research is ongoing, with current efforts aimed at developing a complete set of C++ routines within Perry Cook's real-time synthesis environment, Synthesis Toolkit.

The performance flexibility offered by current real-time woodwind computer models is generally uncontrollable within the context of existing MIDI wind controllers. A new wind controller which allows variable tonehole closure and non-traditional fingering is in the design stages.

References

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6.2.8 Spectral Modeling Synthesis (SMS)

Xavier Serra

Spectral Modeling Synthesis (SMS) is a set of techniques and software implementations for the analysis, transformation and synthesis of musical sounds. SMS software implementations were first done by Xavier Serra and Julius Smith at Stanford University, and more recently by the first author and the music technology group of the Audiovisual Institute of the Pompeu Fabra University in Barcelona. The aim of this work is to get general and musically meaningful sound representations based on analysis, from which musical parameters might be manipulated while maintaining high quality sound. These techniques can be used for synthesis, processing and coding applications, while some of the intermediate results might also be applied to other music related problems, such as sound source separation, musical acoustics, music perception, or performance analysis.

Our current focus is on the development of a general purpose musical synthesizer. This application goes beyond the analysis and resynthesis of single sounds and some of its specific requirements are:

1. it should work for a wide range of sounds;
2. it should have an efficient real time implementation for polyphonic instruments;
3. the stored data should take little space;
4. it should be expressive and have controls that are musically meaningful;
5. a wide range of sound effects, such as reverberation, should be easily incorporated into the synthesis without much extra cost.

The implementation of these techniques has been done in C++ and Matlab, and the graphical interfaces with Visual C++ for Windows 95. Most of the software and the detailed specifications of the techniques and protocols used are publicly available via the SMS Web site[4].

6.2.9 FFT-Based DSP and Spectral Modeling Synthesis

Julius Smith

The Fast Fourier Transform (FFT) revolutionized signal processing practice in the 1960s. Today, it continues to spread as a practical basis for digital systems implementation. Only very recently has it become cost-effective to use the short-time FFT in real-time digital audio systems, thanks to the availability of sufficiently powerful, low-cost, single-chip solutions.

In the digital audio field, FFT-based techniques are ripe to appear in digital mixing consoles, post-production editing facilities, and top-quality digital audio gear. Many music and digital audio "effects" can be conveniently implemented in a unified way using a short-time Fourier analysis, modification, and resynthesis facility.

In the music synthesis field, obtaining better control of sampling synthesis will require more general sound transformations. To proceed toward this goal, transformations must be understood in terms of what we hear. The best way we know to understand a sonic transformation is to study its effect on the short-time spectrum, where the spectrum-analysis parameters are tuned to match the characteristics of hearing as closely as possible. Thus, it appears inevitable that sampling synthesis will migrate toward spectral modeling. Recent developments in constant-Q filterbanks, such as in the wavelet literature, have created new alternatives for consideration. Advanced time-frequency representations, such as the Wigner Distribution, are yielding new insights into time-varying audio spectra.

[4] http://www.iua.upf.es/~sms

In contrast with physical modeling synthesis which models the *source* of a sound, spectral modeling techniques model sound at the *receiver*, the human ear. Spectral modeling is more immediately general than physical modeling since it is capable of constructing an arbitrary stimulus along the basilar membrane of the ear, while new physical models must be developed for each new class of musical instrument. While complex coarticulation effects are more naturally provided by physical models, the short-time Fourier transform can be applied to any sound demonstrating any desired effect to determine what must happen in a spectral sequence to produce that effect.

FFT-based techniques play an important role in (1) the practical implementation of general signal processing systems (fast convolution), (2) advanced effects such as "cross synthesis," time compression/expansion, duration-invariant frequency shifting, and other "phase vocoder" type techniques, and (3) novel synthesis systems based on the direct creation and transformation of spectral events and envelopes.

References

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- Smith, J. O. Music 420 (EE 265) Course Description (see Section 4.1), Stanford University.

6.2.10 Digital Waveguide Modeling of Acoustic Systems

Julius Smith

Digital Waveguide Filters (DWF) have proven useful for building computational models of acoustic systems which are both physically meaningful and efficient for applications such as 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 initially derived to construct digital reverberators out of energy-preserving 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 computational form 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.

In this context, a *waveguide* can be defined as 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 wave impedance of the medium is constant. At changes in the wave impedance, a traveling wave partial transmits and partially reflects in an energy conserving manner, a process known as "scattering." The wave impedance is the square root 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.

Digital 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 DWF structure are equal exactly (at the sampling times and positions, to within numerical precision) to variables propagating in the corresponding physical system. Signal power is defined instantaneously with respect to time and space (just square and sum the wave variables.) 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 DWF system. Because waveguide filters can be specialized to well studied lattice/ladder digital filters, it is straightforward to realize any digital filter transfer function as a DWF. Waveguide filters are also related to "wave digital filters" (WDF) which have been developed primarily by Fettweis. Using a "mesh" of one-dimensional waveguides, modeling can be carried out in two and higher dimensions. In other applications, the propagation in the waveguide is extended to include frequency dependent losses and dispersion. In still more advanced applications, nonlinear effects are introduced as a function of instantaneous signal level.

Digital waveguide filters can be viewed as an efficient discrete-time "building material" for acoustic models incorporating aspects of one-dimensional waveguide acoustics, lattice and ladder digital filters, wave digital filters, and classical network theory.

References

- Music 421 (EE 266) Course Description
- J. O. Smith, "Physical Modeling Synthesis Update", **Computer Music Journal**, vol. 20, no. 2, pp. 44–56, Summer, 1996.
- J. O. Smith, "Physical Modeling Using Digital Waveguides", **Computer Music Journal**, (special issue on physical modeling of musical instruments, part I), vol. 16, no. 4, pp. 74–91, Winter, 1992. A 43-page tutorial based on this material, "Acoustic Modeling Using Digital Waveguides", is now published as Chapter 7 of **Musical Signal Processing**, edited by Roads, Pope, Piccialli, and De Poli; and published by Swets and Zietlinger, the Netherlands, pp. 221-263, 1997.

6.2.11 Signal Processing Algorithm Design Stressing Efficiency and Simplicity of Control

Timothy Stilson

This project deals with the design of digital filters, oscillators, and other structures that have parameters that can be varied efficiently and intuitively. The main criteria for the algorithms are:

- **Efficiency:** The algorithms are intended to be as efficient as possible. This constraint is weighted very high in design decisions.
- **Non-Complexity of Controls:** As a large part of efficiency, the amount of processing that must be done on an input control to make it useful for the algorithm should be minimized. As an example, some filter may have "center frequency" as a control input, but may actually go through a bunch of expensive calculations to turn it into some lower level coefficients that are actually used in the filter calculation. On the other hand, another filter may have design whereby center frequency goes directly into the filter with little change, and the filter uses it in a rather simple calculation (i.e. the ugly math hasn't simply been absorbed into the filter). This constraint often influences the choice of basic algorithms, but also influences the control paradigms. For example, some algorithms may turn out to be vastly more efficient if given some variation of frequency as an input, say period, or log(frequency). In order to remain efficient, the control paradigm may also need to change (the whole system may use period rather than frequency, for example), otherwise there will need to be excessive parameter conversions, which violate the control complexity criterion.
- **Intuitiveness of Controls:** As alluded to in the previous item, certain forms of controls can be more efficient than others. Unfortunately, some efficient parameters may be hard to use for an end-user, i.e. a musician will likely prefer to specify center frequency to a filter algorithm rather than filter coefficients. In order to make algorithms usable, one must either introduce parameter conversion procedures (inefficient) or look for an algorithm that has the desired inputs yet is more efficient.

Often, one decides that a certain amount of inefficiency is livable, and in cases where a parameter changes only rarely, large amounts of inefficiency can be tolerated. But when a parameter must change very often, such as in a smooth sweep or a modulation, inefficiency is intolerable.

In this project, the main application is the field referred to as "Virtual Analog Synthesis", which tries to implement analog synthesis algorithms (in particular, subtractive synthesis) in digital systems. Characteristics of many analog patches were the blurring of the distinction between control signals and audio signals, such as in modulation schemes, or the ability to dynamically (smoothly) control any parameter. Both of these abilities require parameters to change at very high rates, even as fast as the sampling rate. Thus the necessity for efficiently controllable algorithms.

Two subprojects within this project are currently under being researched. First: the design and implementation of an efficient signal generator which generates bandlimited pulse trains, square waves, and sawtooth waves. The algorithm is being designed for basic efficiency, along with considerations for efficient variation of the main parameters: frequency and duty cycle.

Secondly, the connections between control-system theory and filter theory are being explored. One particular avenue of research is the application of Root-Locus design techniques to audio filter design. Root Locus explores the movement of system (filter) poles as a single parameter changes. Certain patterns in root loci appear repeatedly, and can be used in audio filter design to get various effects. A good example is the Moog VCF, which uses one of the most basic patterns in root-locus analysis to generate a filter that has trivial controls for both corner frequency and Q. Several other families of sweepable digital filters based on root-locus have already been found. A particular goal is to find a filter family that efficiently implements constant-Q sweepable digital filters (a problem that, it turns out, is particularly simple in continuous time — the Moog VCF — but is quite difficult in discrete-time).

6.2.12 Applying Psychoacoustic Principles to Soundfield Reconstruction

Steven Trautmann

Simulations and simple experiments have indicated that a broad class of musical signals can benefit from some simple processes aimed at reproducing a soundfield's perception accurately through loudspeakers. These processes attempt to recreate relative phase and amplitude information accurately at the listeners' ears, while allowing distortions elsewhere. The net effect should be to give a more accurate reproduction of important cues for localization and other factors to the listeners. Current work is geared toward expanding these results, by increasing the mathematical rigor and creating further generalizations, and by looking at how other psychoacoustic effects such as masking effects can be applied to further increase the accuracy of the reproduced soundfield's perception.

6.2.13 Identification of Vocal Tract Parameters for the Analysis of Speech and Singing

Yoon Kim

The purpose of this research is to investigate new methods of extracting features in speech that might effectively describe the various aspects of speech and singing, and their relationship to acoustic and articulatory parameters of the vocal tract. In the past, researchers in the signal processing field only focused on speech features that are based on spectral (acoustic) information e.g. formants, linear predictive coding). This was because these features were utilized in applications such as speech compression and recognition, where computation of the parameters was as crucial to the problems as the perceptual aspects. Of course, despite being easily computed from Fourier analysis, these spectral parameters have an inevitable limitation, as they cannot describe the details of the mechanisms behind speech production.

To gain deeper knowledge about the infrastructure of speech production, we need to investigate the extraction of more sophisticated parameters. Previous studies were performed at the articulatory level. However, it is difficult to collect reliable data due to physical limitations of measuring devices and the acoustic coupling of the vocal tract. Thus, alternative methods of obtaining useful information from tractable parameters is in order. The research will thus focus on devising accessible features that provide useful insight on the articulatory dynamics and the physical nature of the vocal tract. These features may provide a basis for solving various problems in speech and singing where a detailed analysis of the interactions between articulatory and acoustic aspects of speech production is required (e.g. pronunciation scoring of speech, diagnosis of singing problems).

6.3 Controllers for Computers and Musical Instruments

6.3.1 Real-time Controllers for Physical Models

Chris Chafe

The computational reductions brought about by new work in algorithms such as the Waveguide filter formulations, along with improvements in DSP chips and other signal processing hardware, have made possible the real-time synthesis of music by physical modeling. Such instruments require new modes and levels of control. Work in increasing the bandwidth of synthesizer control by exploiting all available degrees of freedom has yielded a number of experimental hybrid controllers (Cook, Chafe). Controllers based on the paradigms of wind and stringed instruments have improved the control of models based on these families, and research is being conducted to create a more general controller which is not constrained to a particular family of instruments.

Mapping physical gestures to a DSP synthesis controller is being studied by experimentation. Early studies in simulation (Chafe, 1985) suggested that linear mappings are not the way to go. The current development system allows trial-by-feel investigation of alternative scalings.

The area of tactile feedback (Chafe) is being investigated, as this is an important area of control for the traditional instrument player. Initial trials have begun using actuators feeding audio to the touch point. A general preference has been shown with the technique. The next stage will be to quantify what enhancement, if any, results from feeling the instrument's vibrations. Also, such considerations as tactile frequency bandwidth and vibrations characteristic of contact points will be studied.

New pieces are being written using real-time controllers and the DSP-based physical models. *El Zorro* is a recent composition by Chris Chafe that employs a Lightning Controller (by Buchla and Assòciates). The soloist is steering note-generation algorithms in terms of tempo, tessitura and "riff-type." Gesture and position is tracked with the Lightning's infra-red controllers. Some direct control is exercised over DSP effects via MIDI. A composition project in the works uses the Celletto (an electronic cello) to interact with the DSP synthesis at the control level. The cellist will evoke synthesis related to natural cello behavior directly from the instrument. For example, bow speed might translate into breath pressure control of a wind synthesis.

6.3.2 The Touchback Keyboard

Brent Gillespie

For well over a decade, Max Mathews, John Chowning, George Barth, and others at CCRMA have envisioned a synthesizer keyboard with programmable touch-response—a keyboard with the feel of a grand piano, or, at the push of a button, a harpsichord, a piano-forte or perhaps an altogether new keyboard instrument. Such a keyboard would mitigate the deficiencies in touch-response of present-day synthesizer keyboards. But further, a keyboard with programmable feel could be used to investigate the role of touch-response in the relationship between instrument and musician. For according to the experience of musicians, an instrument's touch-response has a great deal to do with its potential as a musically expressive device. Thus the Touchback Keyboard project has been launched at CCRMA: primarily to provide a means to explore the role of the feel of an instrument in musical expression.

A motorized keyboard of seven keys has been designed and built. It features a central bearing-mount shaft with staggered capstans and off-the-shelf low inertia brushed DC motors. In its unpowered state, each key has the approximate inertia of a standard wooden key. When powered, each key may be made to take on the mechanical impedance characteristic of a key interacting with a virtual whippen, jack, hammer, and damper. Creation of the appropriate mechanical impedance is accomplished through real-time simulation of a dynamical model of the piano action. Particularly salient to the feel at the key are changes in kinematic constraint or changing contact conditions among the wooden and felt parts of the piano action. A full software environment for real-time simulation of systems with changing kinematic constraints has been developed.

The creation of virtual objects which may be touched and manipulated through a motorized device is the central activity of a brand-new field called Haptic Display. The word haptic refers to the tactile and kinesthetic senses, and display highlights the fact that these (essentially robotic) devices are computer interface devices just like a monitor and a loud speaker. Haptic Display draws heavily on the fields of robotics, controls engineering, and psychophysics. It turns out that the handling of changing constraints in dynamical models in a sampled-data setting is of prime interest to the haptic display community.

The Touchback Keyboard now resides at Northwestern University, where it is being used to further investigate the role of touch-response in musical expression. The Touchback Keyboard was used as an open research case-problem in a freshman course in engineering design and communications at Northwestern. Cuong Pham, Philip Tang, and David Zaretsky conducted human subject experiments and contributed a report: "Feel the Music". One experiment involved the role of feel in memorizing instrument identities and another studied skill transfer among instruments with different touch-responses. Two constructs of control theory are being used to inspire and organize further experiments: controllability (as a measure of expressive potential) and observability (as a measure of ease of learning).

See `http://lims.mech.nwu.edu/~brent/` for more information on current work with the Touchback Keyboard.

6.3.3 A New Structure for the Radio-Baton Program

Max V. Mathews

A new structure has been developed for the Radio-Baton program. The new structure unifies the Conductor Program mode of operation and the Improv mode (formerly called the Jaffe-Schloss mode) of operation. With the new structure, the Radio-Baton acts as a pure controller sending triggers and baton position information to the computer that is speaking to it, no matter what kind of computer that may be. The Conductor Program or any Improv program is entirely in the computer, so neither the Radio-Baton nor the MIDI connections need be changed going from one program to another. Also, any computer that speaks MIDI–PC's, Mac's, or Unix platforms–can be used. The new structure has the following advantages:

1. All communications going in both directions between the baton and computer is via standard midi commands (control changes or any three byte midi commands can be used). No system exclusive messages are needed.

2. Each midi command is logically complete in itself. Two or more successive commands are never required to be put together in order to logically complete a message.

3. The baton acts only as a simple controller. All programming for either the Improv mode or the Conductor program mode is done in the computer. Thus programs can be easily revised without burning new eproms.

4. The baton program has only one "state". No command from the computer can be misinterpreted by the baton because it is in the wrong state.

5. Midi data going to and from computer and baton goes over completely separate cables from the information going to the synthesizer. Thus there is no possibility of the synthesizer misinterpreting commands meant for the baton or visa-versa. Also, the synthesizer can use all midi channels–no channels need be reserved for the baton.

Position and Trigger Information Sent from Baton to Computer

The baton will send trigger and position information to the computer encoded as key pressure midi commands as follows:

Information	MIDI Command (3 Bytes)
trigger from stick 1 and whack strength	A0 1 WHACK
trigger from stick 2 and whack strength	A0 2 WHACK
trigger from B14+ button	A0 3 0
trigger from B15+ button	A0 3 1
down trigger from B14- foot switch	A0 3 2
up trigger from B14- foot switch	A0 3 3
down trigger from B15- foot switch	A0 3 4
up trigger from B15- foot switch	A0 3 5
pot 1 current value	B0 4 POT1
pot 2 current value	A0 5 POT2
pot 3 current value	A0 6 POT3
pot 4 current value	A0 7 POT4
stick 1 x current position	A0 8 X
stick 1 y current position	A0 9 Y
stick 1 z current position	A0 10 Z
stick 2 x current position	A0 11 X
stick 2 y current position	A0 12 Y
stick 2 z current position	A0 13 Z

In the default setting pot position information is sent 10 times per second and stick information is sent 50 times per second, the total data rate for position information will be 1020 bytes per second or about 1/3 of the midi channel capacity. This can be reduced if desired.

The data rate for trigger bytes will be much smaller than the data rate for position bytes. However, timing is more important for triggers, so the trigger information will be given priority by the baton.

General Structure of Computer Program

The general structure of the computer program will simply be 1. to use the position information to update and keep current a set of memory locations showing the current values of stick and pot positions; and 2. to execute appropriate functions when it receives triggers. The program must also have a good clock (a millisecond clock) so it can measure the times triggers occur and can schedule events to happen in the future.

Commands from Computer-to-Baton

The computer will need to send the following commands to the baton:

Function	Command from Computer-to-Baton	Response from Baton-to-Computer
test-baton and midi ok	A0 14 0	A0 14 0
turn on position reporting	A0 14 1	1020 bytes/sec data
turn off position reporting	AO 14 2	none
set stick levels	A0 14 3	none
set center stick 1	A0 14 4	none
set center stick 2	A0 14 5	none
increase z sensitivity	A0 14 6	none
decrease z sensitivity	A0 14 7	none
increase x-y sensitivity	A0 14 8	none
decrease x-y sensitivity	A0 14 9	none
increase position int 5 ms	A0 14 10	none
report value in buf[j]	A0 15 j	12 bytes encoding value in buf[j]

Originally, I put the Conductor Program in the Radio-Baton because I doubted that the computers of that era would be fast enough to play a complex score in real-time. Such doubts are no longer appropriate. Tests with the new Baton showed that even a 486 processor running at 30MHz could handle the most complex score in the present Conductor Program repertoire – Beethoven's 5th symphony.

6.3.4 Haptic User Interfaces for the Blind

Sile O'Modhrain and Brent Gillespie

Advances in graphic output technology have opened the window for the development of advanced graphical user interfaces making computers increasingly inaccessible to the blind. To date, developers seeking to overcome this situation have relied on two methods of outputting information: sound and braille. Neither of these have been able to provide an adequate substitute for graphics. For certain applications, such as synthesizer controllers and digital music editing, speech output would conflict with the audio output of the system. Therefore we feel it is necessary to explore other ways of presenting information in a tactile form. Because haptic displays can, like graphics, create virtual objects, they present a more natural analogue than text (as in speech or braille). For example, a motorized mouse can define a button to be felt as well as seen–imagine that a particular area of the mouse pad has a different texture. This force reflecting system would also usefully supplement graphical user interfaces for sighted users.

With support from the Stanford Office of Technology and Licensing we have built such a powered mouse, which we call the Moose. Using this device, we have developed a prototype haptic user interface for Windows 3.1 and we have also experimented with rendering haptically the spectrograms of soundfiles. Those who have used this haptic prototype agree that we have begun to tap a very promising resource, a fact which is reinforced by the sudden increase in commercially available haptic display devices. Therefore we feel it won't be long until haptic display will become a viable component of standard computer systems and blind computer users will have access to applications such as sound editors and MIDI sequencers for which speech output is extremely inadequate but for which haptic output is well suited.

6.4 Psychoacoustics and Cognitive Psychology

6.4.1 Neural Network Models of Musical Cognitive Activities

Jonathan Berger, Daniel Lehmann, and Dan Gang

Artificial neural networks provide a flexible environment within which we model the mechanics and implied associated cognitive processes involved in human prediction of time ordered sequential musical elements. We model an experientially trained listener's cognition of functional tonal western music. By interpreting the distribution of output activations of the network as expectations for the next event in the sequence and comparing this to the consequential event, we establish a quantifiable measurement of the degree of realized expectation. The strength and distribution of output activations provide a method for modeling:

1. Schema based theories of cognition.
2. Processes involved in resolving ambiguities and conflicts of schemas and patterns occurring at different structural or hierarchical levels.
3. Dynamic contextualization, that is, how a context is created, adapted, and accepted or rejected as it unfolds in time.
4. Expectational windows - how contexts create both short range and long range predictions. The interaction of short term and long term memory on these processes.
5. The influence of cyclic or metric organizers on pattern extraction and segmentation.

We propose to design and implement a series of experiments to investigate these implications and to refine and develop new connectionist architectures to build these models. Initial experiments with a compact representation of a limited number of musical dimensions will be followed by a more flexible representation incorporating all the multidimensionality, complexity, and intricacies of a complete musical work.

6.4.2 New and Revised Psychoacoustics Textbook

Perry Cook

The lectures and sound examples from CCRMA's Music 151 course, "Psychophysics and Cognitive Psychology for Musicians" have been collected into a book, which is now under publication contract with MIT Press. The book will be published with an included compact disc of sound and code examples. Publication is expected by Fall 1998. Authors include John Chowning, Perry Cook, Brent Gillespie, Dan Levitin, Max Mathews, John Pierce, and Roger Shepard.

An early version without sound examples, typewriter size pages, has been available at the Stanford bookstore.

6.4.3 Absolute Pitch, Absolute Tempo, Absolute Loudness

Daniel Levitin

Broadly speaking, my research is concerned with the psychology of structure and perceptual organization. How does the brain organize the world around us, create categories, and parse a dense perceptual field? To answer these questions, I have been examining principles of visual and auditory perception (how the brain groups basic elements into objects).

More specifically, my current research projects include work on:

- absolute pitch, including issues about learning, etiology, and categorical perception
- circular statistical models for psychological research
- vowel perception
- memory for musical events
- perception of simultaneity of events (intra-modally and cross-modally)
- music perception and Williams' syndrome patients
- tone deafness/tune deafness, dysmelodia, and amusia
- the search for visual perceptual primitives

For more information, please see `http://www-ccrma.stanford.edu/~levitin/research.html`.

6.4.4 Media and Music

Akiko Orita

The development of multimedia works in recent years has gone hand in hand with the development of technology. It has become easier to create such works using not only computer workstations but PCs as well. As the number of TV channels increases, as well as other media such as film, video, PC-Game, and CD-ROM, more materials are needed.

Multimedia works consist of pictures and sound/music. But they often have been treated as the secondary to pictures. The pictures exist first and then composers write music to fit to them. Music and sound in multimedia works have a great effect on their expression. They add more meanings than the picture itself and/or even change the stream of story compared to that without sound.

Two experiments explored audiovisual interactions when perceiving 3 patterns of matching. For the mismatched excerpts, the original relation between the audio and visual tracks was altered with respect to time or content. For the higher level factor, comparison of the results for matched and mismatched conditions implied an intention of balancing audio and visual meaning. Besides, audio meaning had a direct influence on visual meaning, but only for matched stimuli. For the lower level factor, the influence was independent of degree of matching but feeling of time. The study of factor analysis revealed several kinds and levels of audiovisual interaction.

I focused on the progression of both pictures and music. Music used in audiovisual context has an emotional curve that is related to motion and colors of pictures when it is composed as BGM. Audio and visual context will make stream hand in hand to complete the whole works. Using the software "humdrum" on UNIX to analyze the melody pattern, music, sound and action of pictures are arranged in the same time table. Then there I find the relation between audio and visual materials.

6.5 Machine Recognition in Music

6.5.1 Statistical Pattern Recognition for Prediction of Solo Piano Performance

Chris Chafe

The research involves modeling human aspects of musical performance. Like speech, the exquisite precision of trained performance and mastery of an instrument does not lead to an exactly repeatable performed musical surface with respect to note timings and other parameters. The goal is to achieve sufficient modeling capabilities to predict some aspects of expression in performance of a score. The present approach attempts to capture the variety of ways a particular passage might be played by a single individual, so that a predicted performance can be defined from within a closed sphere of possibilities characteristic of that individual. Ultimately, artificial realizations might be produced by chaining together different combinations at the level of the musical phrase, or guiding in real time a synthetic or predicted performance.

A pianist was asked to make recordings (in the Disklavier MIDI data format) from a progression of rehearsals during preparation of a work (by Charles Ives) for concert. The samples include repetitions of the excerpt from the same day as well as recordings over a period of months. This performance data (containing timing and velocity information) was analyzed using classical statistical feature extraction methods tuned to classify the variety of realizations. Chunks of data representing musical phrases were segmented from the recordings according to an "effort parameter" that has been previously described. Presently under study is a simulation system stocked with a comprehensive set of distinct musical interpretations which permits the model to create artificial performances. It is possible that such a system could eventually be guided in real time by a pianist's playing, such that the system is predicting ahead of an unfolding performance. Possible applications would include present performance situations in which appreciable electronic delay (on the order of 100's of msec.) is musically problematic.

6.6 Historical Aspects of Computer Music

6.6.1 Impact of MIDI on Electroacoustic Art Music

Alex Lane Igoudin

The survey which laid the groundwork for the study was conducted by the author in 1996. Forty-five composers from 13 countries in America, Asia, Australia and Europe, including both coasts of the U.S., were interviewed in the course of the project. The chosen respondents had been active in the field before and after introduction of MIDI regardless of their degree of involvement with the MIDI-based tools. The results of the study accurately reflect the attitudes and experiences of the sampled group of composers. The methods used for conducting the study make it very likely to encounter the same trends existing in the entire possible population.

The study was published as the author's doctoral dissertation at CCRMA, Stanford University. It is available in print from CCRMA and can also be viewed as a postscript file on CCRMA's WWW site. Readers interested in evolution of our field over the last two decades are encouraged to acquaint themselves with the full text of work as it presents a wide panorama electroacoustic art music composers' lore: methodologies, ideas, and practices.

The interaction between art and technology comes to a particularly intense point in the studied case. A new generation of tools led to extinction of previous media for electroacoustic composition and produced wide-ranging reactions from its users and numerous effects on methodology and artistic results. The survey's results exposed complex matrix of reception to the new phenomenon and also presented a diverse panorama of existing compositional methodologies and practices.

The composers' reception of MIDI tools was always a compromise between demands of the individual style and advantages and limitations of the MIDI equipment. Advantages of the protocol (its real-time communication, compatibility between the tools, control capabilities and precision) contrasted its limitations (event-oriented paradigm, low data transfer rate, fixed scales of values and one-way communication limited in the number of channels). The features of the protocol were implemented into the design of the MIDI instruments and combined with other technologies, not directly related to MIDI. Often the same feature could be both limiting to one composer and beneficial to another. In some cases the limitations of MIDI equipment and satisfaction of working with non-MIDI environments has led to the total exclusion of MIDI from the compositional setup. Control over the development of continuous processes, a staple in pre-MIDI electroacoustic music, is particularly problematic with MIDI. The technological tradeoff made for the sake of enhanced user-friendliness and affordability in the larger commercial market limited synthesis capabilities and access and therefore disappointed some composers. However, one can see the emergence of new methods, new practices and new performance solutions that were not present in the pre-MIDI era.

The relative democratization of electroacoustic music is clearly one of the positive effects of MIDI revolution. The affordability of the new set of tools led to the appearance of home computer/electroacoustic music studios. MIDI also had a positive effect on concert practice. Also, MIDI marked the beginning of active commercialization of the field.

About half of the surveyed composers had practiced some kind of live (non-tape) music before MIDI. MIDI gave a boost to this genre, providing reliable, portable, storable devices and connections and raising the number of composers involved into live interactive music. Meanwhile, tape pieces have continued to be the principal performance genre among the art composers just as software synthesis continued to be the major source of timbres after the introduction of MIDI. The evaluation of these preset synthesized sounds in MIDI instruments is unfavorable. In particular, the opinion on the quality of acoustic simulation in such sounds is utterly negative.

As our study has shown, the influence of MIDI is multifaceted. The conflict between the origins of MIDI and the pre-existing compositional practice has not been entirely solved. Instead the results of this investigation show the incorporation of the new tools into the existing tradition, compromise in some elements of interaction, rejection of others and development of new practices.

6.7 Computer Assisted Music and Acoustics Research

6.7.1 The Center for Computer Assisted Research in the Humanities (CCARH)

The Center for Computer Assisted Research in the Humanities (CCARH), located in the Braun Music Center, Stanford University, is concerned with the development of data resources and software applications for music research and allied areas of humanities study and with various teaching and research functions. Its official address is:

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fax (650) 725-9290
Web: `http://ccrma-www.stanford.edu/CCARH/`

Teaching and Research Functions

CCARH introduced a two-quarter graduate sequence on Musical Information and Music Representation to the Stanford curriculum in 1996-97. This course sequence is team-taught by Eleanor Selfridge-Field and Walter B. Hewlett. With its move to Stanford premises in 1996-97, CCARH also established a new computer laboratory in the Braun Music Center. This lab is optimized for dedicated music applications, principally in musical notation and analysis.

Databases

The Center has been engaged since its founding in 1984 in the development of "fulltext" databases of standard musical repertory. Collectively these are known are MuseData(TM).

The CCARH databases are full electronic scores of standard repertory. The original sources are encoded as intelligent information that is extended to support graphical, sound, and analytical applications. An extensive description is found in Volume 9 (1993-94) of Computing in Musicology.

MuseData(TM) databases are device-independent. That is, all of the data are stored in ASCII representations. The database codes, file structures, and operating software have been developed by Walter B. Hewlett. Technical information is available in a separate publication from CCARH. To facilitate applications, works represented in MuseData(TM) code are translated into other application-specific codes. At the present time the application-specific codes supported are MIDI, SCORE, DARMS, and Kern. MIDI supports sound and sequencer applications in multiple platforms. SCORE and DARMS support printing applications on DOS microcomputers. Kern supports analytical applications under UNIX and UNIX simulation software (such as the MKS Toolkit, running under DOS). Other openly documented codes may be supported in the future.

CCARH currently intends to make much of its data available via the WorldWide Web.

Publications

An important corollary to the development of the databases is the assembly of documentation for using the various application-specific codes. *Beyond MIDI: The Handbook of Musical Codes*, edited by Eleanor Selfridge-Field, was published by MIT Press in August 1997. Containing the work of more than 40 authors, it is a comprehensive source covering codes for representing sound, notation, analysis, and interchange of musical data.

Another important aspect of the Center's activity has been in the dissemination of information about current applications in music research. This is reported in the yearbook Computing in Musicology. Volume 10 (1995-96) focuses on analytical software. Volume 11 (in press as of 9/9/97) focuses on melodic analysis and comparison.

CCARH contributed the article "Musical Information in Desktop Publishing" to the IEEE CD-ROM "Standards for Computer-Generated Music" published by the IEEE Computer Society Press in HTML and PDR formats in 1996.

Invited visitors

- David Huron, developer of the Humdrum Toolkit for Musical Analysis and newly appointed professor of Music and Cognition at the Ohio State University, was in residence in 1996-97. His stay enabled him to complete a user guide for the Humdrum community, to offer a summer course on

Humdrum, to develop a database of musical themes searchable on the World-Wide Web, to further develop conversion software from the MuseData format to the Kern format used by Humdrum, and to participate in various teaching and research activities.

- Andreas Kornstaedt, a graduate student in software engineering at the University of Hamburg (Germany), spent 1996-97 in residence at CCARH. He was involved in developing a user interface for thematic searches, in developing CCARH's Web capabilities, and in creating an enriched Kern format that preserves page-description information used by the SCORE notation program. Mr. Kornstaedt wrote a conversion program from the SCORE binary file format to this enriched version of Kern and assisted Dr. Huron in many of his activities.
- Douglas R. Hofstadter, the well-known author of Goedel, Escher, Bach and numerous other books concerned with issues of cognition, organized a lecture series on computer challenges to human creativity for the Autumn quarter of 1997-98. Prof. Hofstadter is on leave from Indiana University.

6.7.2 The Musical Acoustics Research Library

Gary P. Scavone and **Max V. Mathews**

The Musical Acoustics Research Library (MARL) is a collection of independent archives or libraries assembled by distinguished groups or individuals in the field of musical acoustics research. MARL is directed by representatives of each member library, in conjunction with the Center for Computer Research in Music and Acoustics (CCRMA) at Stanford University, which maintains the contents of each library. Currently, MARL is comprised of the Catgut Acoustical Society Library, the Arthur H. Benade Archive, and the John Backus Archive.

Background and History

In 1990, the Catgut Acoustical Society (CAS) began a search for a permanent home for its extensive files and research library. In 1992, the Center for Computer Research in Music and Acoustics (CCRMA) at Stanford University was selected as the site for this library.

In conjunction with the establishment of the Catgut Acoustical Society Library, CCRMA was working to obtain the personal archives of Arthur Benade and John Backus. Benade was a physicist working at Case Western Reserve University and Backus was a physicist working at the University of Southern California. Both were world leaders in the study of wind instrument acoustics. The personal files of John Backus were acquired in 1995. An agreement for the establishment of the Arthur H. Benade Archive at CCRMA was reached in 1997.

In the Fall of 1996, representatives of CAS and CCRMA, together with Virginia Benade, established the Musical Acoustics Research Library at CCRMA in order to provide a single point of reference for the various collections.

Purpose and Goals

The Musical Acoustics Research Library at CCRMA has been established for the purpose of preserving and maintaining a complete and up-to-date repository of knowledge on musical acoustics. CCRMA is committed to honoring requests for information from interested parties all over the world, and further to incorporate new knowledge as it becomes available. It is intended that the collection be made as accessible as possible and its existence be made known around the world. To this end, World Wide Web pages (`http://www-ccrma.stanford.edu/CCRMA/Collections/MARL/`) have been created to outline the contents of the library. Further, documents from the library will be made available, via the Web pages, in Portable Document Format (PDF) on a "per request" basis.

Activities

The process of transferring the CAS library to CCRMA and cataloging its contents is currently underway. Likewise, the process of documenting and transferring the Arthur H. Benade archive has begun. CCRMA has recently acquired and configured the Adobe Capture software, which allows library documents to be scanned, converted to PDF files, and made available over the Internet.

7 Recordings

Recordings of works realized at CCRMA include the following:

- *Computer Music @ CCRMA vol. I.* Works of Aguiar, Norton, Lopez-Lezcano, Hopkins, Hind, and Roy. Produced by Celso Aguiar and Marco Trevisani at CCRMA, Stanford University, 1997.
- *Computer Music @ CCRMA vol. II.* Works of Trevisani, Landa, Soley, Edwards, Dong, and Brummer. Produced by Celso Aguiar and Marco Trevisani at CCRMA, Stanford University, 1997.
- *Center for Computer Research in Music and Acoustics,* CDCM, Computer Music Series, Vol 8. Works of Chafe, Schindler, Jaffe, Berger, and Morrill. Digitally mastered CD, CRC2091, Centaur Records, 1991.
- *John Chowning,* – Phoné, Turenas, Stria, and Sabelithe. Digital compact disk, WER2012-50 Wergo, Germany, 1988, Harmonia Mundi, Distributors.
- *Computer Music from CCRMA, vol. I.* Digitally mastered cassette with works by Moorer, Schloss, Schottstaedt, Chafe, Jaffe, Berger, and Chowning. Produced by Jan Mattox for CCRMA, 1982.
- *Computer Music from CCRMA, vol. II.* Digitally mastered cassette with works by various composers, 1984 (out of print).
- *Current Directions in Computer Music - Sound Examples.* Digital compact disk to accompany the book *Current Directions in Computer Music,* J. R. Pierce and M. V. Mathews, editors, MIT Press, Cambridge, MA, 1989.
- *Dinosaur Music.* Digital compact disk. Jaffe – "Silicon Valley Breakdown," Chafe – "In A Word," Schottstaedt – "Water Music I & II," and "Dinosaur Music," WER2016-50, Wergo, Germany, 1988, Harmonia Mundi, Distributors.
- *Intercambio Exchange.* Digital compact disk containing computer music from Buenos Aires and California. Works of Cetta, Chafe, Dobrian, Willey, Losa, Krupowicz, Calzon, Lopez-Lezcano, Pozzati, Jaffe, Labor, Cerana, Linan, Lyon and Viera. Produced by CCRMA, LIPM, and CRCA, 1994.
- *Computer Music Journal Volume CD,* – digital compact disk to accompany the 20th Anniversary Issue includes Chowning – "Turenas", MIT Press, Cambridge, MA, 1996.
- *The Virtuoso in the Computer Age-V.* Music for Mathews/Boie Radio-Drum and Radio Baton, CDCM Vol. 15. CRC 2190 Centaur, 1994. Works by Jaffe ("Terra Non Firma"), Jaffe/Schloss ("Wildlife"), Radunskaya ("A wild and reckless place"), Austin ("Mathews Epsiode"), and Appleton ("Pacific Rimbombo").
- *New Music for Orchestra.* VMM 3024, Vienna Modern Masters, 1994. Works by Jaffe ("Whoop For Your Life!") and others.
- *The Digital Domain.* Elektra/Asylum Records 9 60303-2, 1983. Works by Jaffe (Finale to "Silicon Valley Breakdown"), McNabb ("Love in the Asylum"), Schloss ("Towers of Hanoi"), Mattox ("Shaman"), Rush, Moorer ("Lions are Growing"), and others.
- David Jaffe. *The Seven Wonders of the Ancient World,* – digital compact disk available in September, Well-Tempered Productions, 1996.
- David Jaffe. *XXIst Century Mandolin - acoustic and computer music for the mandolin.* WTP5164, Well Tempered Productions, 1994, Allegro Records, Distributors.

- Michael McNabb. *Computer Music.* Digitally mastered LP. McNabb – "Dreamsong," "Love in the Asylum," "Orbital View," (LP out of print) - CD now available as WER-2020-2, Wergo, Germany, 1994, Harmonia Mundi, Distributors.
- Michael McNabb. *Invisible Cities.* Digital compact disk, WER-2015-50, Wergo, Germany, 1988, Harmonia Mundi, Distributors.
- *Musica Maximalista - Maximal Music Vol. 2,* – digital compact disk containing Celso Aguiar – "Piece of Mind", CD MM-002, Studio PANorama, Brazil, 1996.
- *Musica Maximalista - Maximal Music Vol. 3,* – digital compact disk containing Celso Aguiar – "All blue, I write with a blue pencil, on a blue sky", CD MM-003, Studio PANaroma, Brazil, 1997. CD of the II International Electroacoustic Music Competition of So Paulo.
- *II SBCM (II Brazilian Symposium on Computers and Music),* – digital compact disk containing Celso Aguiar – "Piece of Mind", DISC MFG., INC., BHS1046, Brazil, 1996.
- *The Science of Musical Sound - Musical Examples.* Cassette tape produced by Jan Mattox for CCRMA to accompany the book *The Science of Musical Sound,* J. R. Pierce, Scientific American, 1985.
- *Unknown Public (04): Musical Machinery,* – digital compact disk containing Nicky Hind – "Rain", Unknown Public, UPCD04, United Kingdom, 1994.

For availability of a particular recording on the list, please contact the composer.

8 Publications

The following is a list of publications since 1996 by people from CCRMA. An extensive list of publications since 1970 is available online at www-ccrma.stanford.edu/Overview/publications.html. A printed list of CCRMA publications from 1970 – 1995 is available from CCRMA as Stanford University Department of Music Technical Report STAN-M-103. Stanford University Department of Music Technical Reports are available from CCRMA. Publications with notated prices are also available from CCRMA.

Berger, J. and Gang, D. (1997). A neural network model of metric perception and cognition in the audition of functional tonal music. In ICMC (1997). (Also contained in STAN-M-101).

Chafe, C. (1997). Statistical pattern recognition for prediction of solo piano performance. In ICMC (1997). (Also contained in STAN-M-101).

Chafe, C. and O'Modhrain, S. (1996). Musical muscle memory and the haptic display of performance nuance. In ICMC (1996). (Also contained in STAN-M-99).

Cook, P. R. (1996). Singing voice synthesis: History, current work, and future directions. *Computer Music Journal*, 20(3).

Gillespie, B. (1996). *Haptic Displays of Systems with Changing Kinematic Constraints: The Virtual Piano Action.* Ph.D. thesis, Dept. of Mech. Eng. Stanford University. Available as Stanford University Department of Music Technical Report STAN-M-92 ($12.00).

ICMC (1996). *Proceedings of the 1996 International Computer Music Conference, Hong Kong.* International Computer Music Association.

ICMC (1997). *Proceedings of the 1997 International Computer Music Conference, Thessaloniki, Greece.* International Computer Music Association.

Igoudin, A. and Smith, J. O., editors (1996). *CCRMA Report, May 1996*, Stanford University Department of Music Technical Report STAN-M-98. ($6.00).

Igoudin, A. L. (1997*a*). *Impact of MIDI on Electroacoustic Art Music.* Ph.D. thesis, Dept. of Music, Stanford University. Available as Stanford University Department of Music Technical Report STAN-M-102 ($14.00).

Igoudin, A. L. (1997*b*). Impact of MIDI on electroacoustic art music. In ICMC (1997). (Also contained in STAN-M-101).

Kim, Y., Franco, H., and Neumeyer, L. (1997). Automatic pronunciation scoring of specific phone segments for language instruction. In *Proceedings of EUROSPEECH '97.*

Kunze, T. and Taube, H. (1996). SEE – a structured event editor: Visualizing compositional data in Common Music. In ICMC (1996). (Also contained in STAN-M-99).

Levine, S. (1996*a*). Critically sampled third octave filter banks. In ICMC (1996). (Also contained in STAN-M-99).

Levine, S. (1996*b*). Effects processing on audio subband data. In ICMC (1996). (Also contained in STAN-M-99).

Levine, S. N., Verma, T. S., and Smith, III, J. O. (1997). Alias-free multiresolution sinusoidal modeling for polyphonic, wideband audio. In *Proceedings of the IEEE Workshop on Applications of Signal Processing to Audio and Acoustics, New Paltz, NY*, New York. IEEE Press.

Lopez-Lezcano, F. (1996). PadMaster: banging on algorithms with alternative controllers. In ICMC (1996). (Also contained in STAN-M-99).

O'Modhrain, M. S. (1997). Feel the music: Narration in touch and sound. In ICMC (1997). (Also contained in STAN-M-101).

O'Modhrain, M. S. and Gillespie, R. B. (1997). The moose: A haptic user interface for blind persons. In *Proceedings of the Sixth World Wide Web Conference: Access Track*. This paper differs somewhat from STAN-M-95.

Pierce, J. R. and Duyne, S. A. V. (1997). A passive non-linear digital filter design which facilitates physics-based sound synthesis of highly nonlinear musical instruments. *Journal of the Acoustical Society of America*, 101(2):1120–1126. ($3.00).

Porcaro, N., Putnam, W., Scandalis, P., Jaffe, D., Smith, J. O., Stilson, T., and Duyne, S. V. (1996*a*). SynthBuilder and Frankenstein, tools for the creation of musical physical models. In G. Kramer, editor, *International Conference on Auditory Display, Palo Alto*. Santa Fe Institute and Xerox Parc. Available online at http://www.santafe.edu/ icad/.

Porcaro, N., Scandalis, P., Jaffe, D., and Smith, J. O. (1996*b*). Using SynthBuilder for the creation of physical models. In ICMC (1996). (Also contained in STAN-M-99).

Putnam, W. and Smith, J. O. (1997). Design of fractional delay filters using convex optimization. In *Proceedings of the IEEE Workshop on Applications of Signal Processing to Audio and Acoustics, New Paltz, NY*, New York. IEEE Press.

Putnam, W. and Stilson, T. (1996). Frankenstein: A low cost multi-DSP compute engine for Music Kit. In ICMC (1996). (Also contained in STAN-M-99).

Risset, J.-C. and Duyne, S. A. V. (1996). Real-time performance interaction with a computer-controlled acoustic piano. *Computer Music Journal*, 20(1).

Rocchesso, D. and Smith, J. O. (1997). Circulant and elliptic feedback delay networks for artificial reverberation. *IEEE Transactions on Speech and Audio Processing*, 5(1):51–60.

Scavone, G. P. (1996). Modeling and control of performance expression in digital waveguide models of woodwind instruments. In ICMC (1996), pp. 224–227.

Scavone, G. P. (1997). *An Acoustic Analysis of Single-Reed Woodwind Instruments with an Emphasis on Design and Performance Issues and Digital Waveguide Modeling Techniques*. Ph.D. thesis, Department of Music, Stanford University. Available as Stanford University Department of Music Technical Report STAN-M-100 ($22.00) or from ftp://ccrma-ftp.stanford.edu/pub/Publications/Theses/GaryScavone-Thesis/.

Scavone, G. P., editor (1998*a*). *CCRMA Overview, January 1998*, Stanford University Department of Music Technical Report STAN-M-104. ($6.00).

Scavone, G. P., editor (1998*b*). *CCRMA Publications, 1970 – 1995*, Stanford University Department of Music Technical Report STAN-M-103. ($6.00).

Scavone, G. P. and Smith, J. O. (1997*a*). Digital waveguide modeling of woodwind toneholes. In ICMC (1997), pp. 260–263.

Scavone, G. P. and Smith, J. O. (1997*b*). Scattering parameters for the Keefe clarinet tonehole model. In *Proceedings of the International Symposium on Musical Acoustics (ISMA-97), Edinburgh, Scotland*, pp. 433–438.

Smith, J. O. (1996*a*). Discrete-time modeling of acoustic systems with applications to sound synthesis of musical instruments. In *Proceedings of the Nordic Acoustical Meeting, Helsinki*, pp. 21–32. (Plenary paper.) Available online at http://www-ccrma.stanford.edu/~jos/.

Smith, J. O. (1996*b*). Physical modeling synthesis update. *Computer Music Journal*, 20(2).

Smith, J. O. (1996*c*). Recent results in discrete-time models of musical instruments. In *Proceedings of the Tempo Reale Workshop on Physical Model Synthesis of Sound, Florence*, pp. 1–6. (Keynote paper).

Smith, J. O. (1997*a*). Acoustic modeling using digital waveguides. In C. Roads, S. T. Pope, A. Piccialli, and G. De Poli, editors, *Musical Signal Processing*, pp. 221–263. Netherlands: Swets and Zietlinger. Also available online at http://www-ccrma.stanford.edu/~jos/.

Smith, J. O. (1997*b*). Nonlinear commuted synthesis of bowed strings. In ICMC (1997). (Also contained in STAN-M-101).

Smith, J. O. (1997*c*). Principles of digital waveguide models of musical instruments. In M. Kahrs and K. Brandenburg, editors, *Applications of DSP to Audio & Acoustics*. Kluwer Academic Publishers. In press.

Smith, J. O. and Karjalainen, M. (1996). Body modeling techniques for string instrument synthesis. In ICMC (1996). (Also contained in STAN-M-99).

Smith, J. O. and Rocchesso, D. (1998). Aspects of digital waveguide networks for acoustic modeling applications. Submitted for publication.

Smith, J. O. and Scavone, G. P. (1997). The one-filter Keefe clarinet tonehole. In *Proceedings of the IEEE Workshop on Applications of Signal Processing to Audio and Acoustics, New Paltz, NY*, New York. IEEE Press.

STAN-M-101 (1997). *CCRMA Papers Presented at the 1997 International Computer Music Conference, Thessaloniki, Greece.* Stanford University Department of Music Technical Report, ($6.00).

STAN-M-99 (1996). *CCRMA Papers Presented at the 1996 International Computer Music Conference, Hong Kong.* Stanford University Department of Music Technical Report, ($7.00).

Stilson, T. and Smith, J. O. (1996*a*). Alias-free digital synthesis of classic analog waveforms. In ICMC (1996). Available online at http://www-ccrma.stanford.edu/~stilti/. (Also contained in STAN-M-99).

Stilson, T. and Smith, J. O. (1996*b*). Analyzing the Moog VCF with considerations for digital implementation. In ICMC (1996). Available online at http://www-ccrma.stanford.edu/~stilti/. (Also contained in STAN-M-99).

Stilson, T. S. (1997). Applying root-locus techniques to the analysis of coupled modes in piano strings. In ICMC (1997). (Also contained in STAN-M-101).

Taube, H. and Kunze, T. (1997). An HTTP interface to Common Music. In ICMC (1997). (Also contained in STAN-M-101).

Van Duyne, S. A. (1997). Coupled mode synthesis. In ICMC (1997). (Also contained in STAN-M-101).

Van Duyne, S. A., Jaffe, D. A., Scandalis, P., and Stilson, T. S. (1997). A lossless, click-free, pitchbendable delay line loop interpolation scheme. In ICMC (1997). (Also contained in STAN-M-101).

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Verma, T. S., Levine, S. N., and Meng, T. H. (1997). Transient modeling synthesis: a flexible analysis/synthesis tool for transient signals. In ICMC (1997). (Also contained in STAN-M-101).

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