CENTER FOR COMPUTER RESEARCH IN MUSIC AND ACOUSTICS
DEPARTMENT OF MUSIC, STANFORD UNIVERSITY
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CCRMA OVERVIEW

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EDITED BY
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Contents

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: Composition, 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, 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, visiting researchers 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, including exchange concerts with area computer music centers and 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, Fender Musical Instruments Corporation, Hewlett Packard, IBM Computer Music Center, Interval Research, ITRI CCL Taiwan, Kind of Loud Technologies, Korg, Matsushita, Media Vision, McDSP, NEC, NeXT Computer, Nokia Group, NTT Communication Science Laboratories, 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. 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
brg	Jonathan Berger	Associate Professor of Music
mab	Marina Bosi	Consulting Professor of Music
cc	Chris Chafe	Professor of Music, CCRMA Director
jc	John Chowning	Professor of Music, Emeritus
vibeke	Vibeke Cleaver	Administrative Assistant
n/a	Walter B. Hewlett	Consulting Professor of Music
jay	Jay Kadis	Audio Engineer / Lecturer
nando	Fernando Lopez-Lezcano	System Administrator / Lecturer
mvm	Max V. Mathews	Professor of Music (Research)
cnichols	Charles Nichols	Associate Technical Director
gary	Gary Scavone	Technical Director / Lecturer
bil	William Schottstaedt	Research Associate
tricia	Tricia Schroeter	Administrative Associate
esf	Eleanor Selfridge-Field	Consulting Professor of Music
malcolm	Malcolm Slaney	Lecturer
jos	Julius O. Smith III	Associate Professor, Music and Electrical Engineering
lcs	Leland Smith	Professor of Music, Emeritus
verplank	Bill Verplank	Researcher and Lecturer

2.2 Engineering Graduate Students

Login	Name	Degree Program
dattorro	Jon Dattorro	PhD Electrical Engineering
guille	Guillermo Garcia	PhD Electrical Engineering
arvindh	Arvindh Krishnaswamy	PhD Electrical Engineering
jacobliu	Yi-Wen Liu	PhD Electrical Engineering
asmaster	Aaron Steven Master	PhD Electrical Engineering
vickylu	Hui-Ling Lu	PhD Electrical Engineering
harv23	Harvey Thornburg	PhD Electrical Engineering

2.3 Music Graduate Students

Login	Name	Degree Program
cburns	Christopher Burns	PhD Computer-Based Music Theory and Acoustics
castelli	Luigi Paolo Castelli	MA Science and Technology
pchordia	Parag Chordia	PhD Computer-Based Music Theory and Acoustics
lonny	Lonny Chu	PhD Computer-Based Music Theory and Acoustics
pdelac	Patricio de la Cuadra	PhD Computer-Based Music Theory and Acoustics
gurevich	Michael Gurevich	PhD Computer-Based Music Theory and Acoustics
colfax	Timothy Colfax Hankins	MA Science and Technolgy
pph	Patty Huang	PhD Computer-Based Music Theory and Acoustics
randal	Randal Leistikow	PhD Computer-Based Music Theory and Acoustics
jdmiller	Joel David Miller	MA Science and Technology
unjung	Unjung Nam	PhD Computer-Based Music Theory and Acoustics
carmenng	Carmen Ng	MA Science and Technology
cnichols	Charles Nichols	PhD Computer-Based Music Theory and Acoustics
norton	Jonathan Norton	PhD Computer-Based Music Theory and Acoustics
cotto	Chris Otto	MA Science and Technology
juan	Juan Carlos Pampin	PhD Computer-Based Music Theory and Acoustics
mromaine	Matthew Matsukane Imai Romaine	MA Science and Technology
anrew	Andrew Jorgen Roper	MA Science and Technology
bschiett	Bert Schiettecatte	MA Science and Technology
rsegnini	Rodrigo Segnini	PhD Computer-Based Music Theory and Acoustics
serafin	Stefania Serafin	PhD Computer-Based Music Theory and Acoustics
tamara	Tamara Smyth	PhD Computer-Based Music Theory and Acoustics
leigh	Leigh VanHandel	PhD CCRMA-cology
jhw	Jeff Walters	MA Science and Technology
carrlane	Carr Lane Wilkerson	MA Science and Technology
rswilson	Scott Wilson	PhD Computer-Based Music Theory and Acoustics

2.4 Visiting Scholars

Login	Name	Home Affiliation	Term
bensa	Julien Bensa	Visiting Researcher, France	
jdc	Joanne Carey	Composer, USA	ongoing
jan	Jan Chomyszyn	Psychoacoustic Researcher, Poland	ongoing
mo	Maureen Chowning	Singer/Composer, USA	ongoing
deguchi	Sachiko Deguchi	Visiting Scholar, Japan	
daj	David Jaffe	Composer/Engineer, USA	ongoing
kronland	Richard Kronland-Martinet	Visiting Scholar, France	
peer	Peer Landa	Composer, Norway	ongoing
juanig	Juan Reyes	Composer/Researcher, Columbia	ongoing
ystad	Solvi Ystad	Visiting Scholar, Norway	

2.5 Undergraduate Students

Login	Name	Degree Program
n/a	Gha-is Abduljaami	Music, Science and Technolodgy
n/a	Daniel Patrick Boatman	Music, Science and Technolodgy
dkc	David Kristoffer Chisholm	Music, Science and Technolodgy
danielsm	Michelle Lee Daniels	Music, Science and Technolodgy
trakstar	Kim Harlan Gaston, Jr.	Music, Science and Technolodgy
sandyg	Sanford Mendel Greenfield	Music, Science and Technolodgy
jhorton	John Wayne Horton	Music, Science and Technolodgy
kalomas	Anthony Gabriel Kalomas	Music, Science and Technolodgy
leslie	Grace Leslie	Music, Science and Technolodgy
n/a	Erica Waying O'Young	Music, Science and Technolodgy
requenez	Edward Requenez	Music, Science and Technolodgy
nschuett	Nathan August Schuett	Music, Science and Technolodgy
ods	Owen Duncan Smith	Music, Science and Technolodgy
quasar	Timothy Pearce Stonehocker	Music, Science and Technolodgy
zarrillo	Katerina Michela Zarrillo	Music, Science and Technolodgy

2.6 Collaborators

Login	Name	Affiliation
prc	Perry R. Cook	Assistant Professor, Computer Science and Music, Princeton University
dhuron	David Huron	Professor, School of Music, Ohio State University
levitin	Daniel Levitin	Assistant Professor of Psychology and Music, McGill University
dex	Dexter Morrill	Professor, Composition, Colgate University
xjs	Xavier Serra	IUA - Phonos, Universitat Pompeu Fabra, Barcelona, Spain
hkt	Rick Taube	Assistant Professor, Composition, University of Illinois

2.7 Industrial Affiliates

Company	Address
Digidesign	Palo Alto, CA
Fender Musical Instruments Corporation	Scottsdale, AZ
Kind of Loud Technologies	Santa Cruz, CA
McDSP	Palo Alto, CA
Nokia Group	Helsinki, Finland
NTT Communication Science Laboratories	Kanagawa, Japan
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 multichannel sound for teaching, concerts, and acoustic experimentation, an adjoining control room/studio, a digital multi-track recording studio with adjoining control room, two additional studios with digital editing facilities, 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 underlying network. A gateway connects the network 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 single and dual processor Intel and AMD based PCs running Linux (with some of the older ones still dual-booting Linux and NEXTSTEP), a few Silicon Graphics workstations, NeXT workstations (for old time's sake) and PowerPC Macintosh computers. All machines are connected through a switched high speed backbone and several servers provide shared services and resources to all computers in a way that is transparent to the users. A high speed connection to the Stanford University Network (SUNET) provides connectivity with the rest of the world, including direct access to the new Internet 2 high speed network. Soundfile manipulation and MIDI input and output are supported on all platforms. Digital multichannel playback is supported on some Linux workstations and on the Macs through several Pro Tools systems. Almost all Linux workstations have high quality 24bit/96KHz soundcards installed. Digital audio processors include a Studer-Editech Dyaxis II system, two Digidesign Pro-Tools systems with CD-R drives, digital i/o cards on Linux systems, and several Panasonic DAT recorders. Text and graphics are handled by an HP 4c color scanner on the unix-based systems and by high resolution network connected printers.

The recording studio consists of a control room and an adjoining recording studio. Equipment available currently includes three Tascam DTRS 8-track digital recorders (one DA-78HR and two DA-38s), a Tascam 80-8 1/2" analog 8-track recorder (with dbx), an Ampex ATR-104 analog 1/2" 4-track recorder (with dbx and/or Dolby A), a Mackie Digital Eight Bus (D8B) mixing console, a Presonus M80 eight-channel mic preamp, a Panasonic SV-3800 DAT recorder, a Waves L2 UltraMaximizer, a Lexicon 224XL digital reverberator, an Eventide Orville processor, Westlake BBSM-10 and JBL 4206 monitors, and outboard gear including equalizers, LA-2A and 1176 compressors, and digital effects processors. A Linux PC-based computer system is available in the control room and has a digital multichannel conenection to the mixer. Recorders may be linked together via SMPTE time code, which will also synchronize the Mac sequencer software. Microphones available in the recording studio include a Neumann TLM-193, two AKG C414B/ULSs, two AKG C460s (with interchangeable cardioid and omni capsules), a Beyer M-500, a Sennheiser MD-421, two Sennheiser E604s, two Electrovoice RE-20s, an Electrovoice N/D868, two Shure Beta-57s, and several Shure SM-57s. There is a Yamaha C7 Disklavier MIDI grand piano in the studio.

The MIDI part of Studio C is organized around a PowerMac G4 computer and an Opcode Studio 5 MIDI interface/MIDI patcher. There is a Yamaha KX-88 weighted-key controller and MIDI equipment including Yamaha SY-99 and VL-1 synthesizers, TX-802 module, Korg Wavestation A/D and X3R modules and Wavedrum synthesizer, E-Mu Proteus/2 module and ESI-32 sampler, and Kurzweil K2000R. There is a Yamaha Disklavier upright piano as well. The Studio C audio system includes a Mackie 24-8 analog mixer, Tascam DA-38, Panasonic SV-3700 DAT recorder, Denon DN-600F CD player, and ProTools MIXplus with 888 I/O and many TDM plug-ins. Monitoring is via four JBL LSR-28P powered speakers. Signal processing is available from a Korg A-1 multi-effects processor. A Plextor 8/20 CD writer is part of the studio as well and CD-Rs can be written from Toast and Jam software from files edited in ProTools or Peak programs.

Studio E is a ProTools-based room with some MIDI capability. Audio equipment includes a Tascam DA-88 recorder, Tascam DM-24 digital mixer, and Genelec 1030A monitors. The ProTools system running on a PowerMac G3 features a ProTools—24 core and an 888 I/O module. Several ProTools TDM plug-ins are available and may be shared by Peak software. MIDI equipment includes an E-Mu Emulator IV, Korg X3R, and a Kurzweil K2000 keyboard connected to an Opcode Studio 5LX interface. A Linux workstation is also available with a Midiman Delta 1010 / 1010AI combination providing 8-channel digital I/O to the system.

Studio D is CCRMA's digital editing and 3D sound facility. Equipment available includes a Studer-Editech Dyaxis II digital editing processor running on a PowerMac G3, a Roland VM-7100 digital mixing system and a Z-systems digital patchbay connecting a Tascam DA-88 with TDIF-to-lightpipe converter, a Panasonic SV-3700 DAT recorder, a Denon CD player with digital output, and a Linux workstation with Midiman Delta 1010 / 1010AI digital 8-channel interface. Eight channel monitoring is through Mackie HR824 speakers and stereo monitoring is through Meyer Sound Labs Model 833 loudspeakers.

The CCRMA software has been developed over more than twenty-years, 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 multi-platform environment for real-time DSP research, STK, is being jointly developed at CCRMA and Princeton University. 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 include the MA/MST 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/courses/`).

Courses offered at CCRMA include:

- **Music 120: Introduction to Sonification** (Winter 2002)

  Principles and application development of auditory display of complex data.

- **Music 120: Musique Concrète in the Digital Era** (Fall 2002)

  Introduction to experimental music composition using computer software (Pro Tools). For music majors or non-majors, novice or experienced composers alike; geared toward computer music beginners. Topics include: compositional techniques; sound editing; basic signal processing; stereo and multi-channel diffusion; electronic music performance practice; historical overview of related electronic music; discussion of the meaning of sound, the aesthetic and legal ramifications of plunderphonics, and metaphor in electronic music. Students will complete regular weekly composition etudes and share them via the web. Larger projects, including a work involving live improvisation and a class collaboration, will be presented in concert.

- **Music 150: Musical Acoustics.**

  Elementary physics of vibrating systems, waves, and wave motion. Time- and frequency-domain analysis of sound. Room acoustics, reverberation, and tuning systems. Acoustics of musical instruments - voice, strings, winds, and percussion. Emphasis on practical aspects of acoustics in music making. Hands-on and computer-based laboratory excercises.

- **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 192: Theory and Practice of Recording**
  - **Music 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, microphone selection and placement, grounding and shielding techniques.
  - **Music 192B: Advanced Sound Recording Technology.**
    Topics: digital audio including current media, formats, editing software, post-processing techniques, noise reduction systems, advanced multi-track techniques, dynamic range processing and delay-based effects.
  - **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 250: Computer-Human Interaction Technology**
  - **Music 250A: HCI Theory and Practice.**
    Human-computer interface (HCI) issues as they relate to music applications in composition and performance. Project-oriented, examining issues from the technical and theoretical perspectives of computer science, haptics, and music theory.
  - **Music 250B: HCI Performance Systems.**
    Continuation of 250A, concentrating on interactive computer-music performance systems.
- **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 255: Orchestration and Timbre Analysis.**

  An introduction to timbre analysis methods with emphasis on analysis of formant characteristics of musical instruments and application to orchestration.

- **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. To sign up for the seminar mailing list, send an e-mail request to hearing-seminar-request@ccrma.stanford.edu. Include the word subscribe in the body of that message.

- **Music 320: Introduction to Digital Audio Signal Processing.**

  A first course in digital signal processing for music and audio research. Topics: complex numbers, sinusoids, spectrum representation, sampling and aliasing, digital filters, frequency response, z-transforms, transfer-function analysis, and associated Matlab software. See web site: `http://www-ccrma.stanford.edu/courses/320/`.

- **Music 420: Audio Applications of the Fast Fourier Transform (FFT).**

  Spectrum analysis and signal processing using the FFT, with emphasis on audio applications. Topics: DFT filter bank; Fourier theorems; spectrum analysis parameters; FFT windows; cyclic and acyclic convolution using the FFT; FIR filter design; phase and channel vocoders; the overlap-add and filter-bank-summation methods for short-time Fourier analysis, modification, and resynthesis; sinusoidal modeling; sines+noise+transients modeling; perfect-reconstruction filter banks. See web site: `http://www-ccrma.stanford.edu/courses/420/`. Prerequisite: Music 320 or equivalent.

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

  Computational methods in digital audio effects and sound synthesis based on acoustic models. Topics: sampled traveling waves; acoustic simulation with delay lines, digital filters, and nonlinear elements; comb filters; allpass filters; artificial reverberation and spatialization; delay-line interpolation and sampling-rate conversion; phasing, flanging, and chorus effects; efficient computational models of strings, woodwinds, brasses, and other musical instruments; finite difference schemes; modal synthesis; waveguide meshes; wave digital filters; and virtual analog. See web site: `http://www-ccrma.stanford.edu/courses/421/`. Prerequisites: Music 320 or equivalent.

- **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: Graduate Seminar in Signal Processing Research.**

  Ongoing seminar for graduate students pursuing research in DSP applied to music or audio. See web site: `http://www-ccrma.stanford.edu/courses/423/`.

4.2 Workshops

CCRMA also offers a series of one- or two-week summer workshops open to participants outside the Stanford community. Information regarding courses to be offered during the coming summer can be accessed at `http://www-ccrma.stanford.edu/`. Courses offered during the last few summers have included the following:

- **Linux Sound: Open Source Music Synthesis, Composition, and Audio Programming**

  CCRMA has been using the Linux operating system for music composition, synthesis, and audio DSP research since 1996. This workshop will focus on currently available open source tools and environments for computer music research and composition using Linux. The workshop will include an overview of some of the most popular linux distributions and a brief installation clinic with specific focus on audio, midi and real-time performance (dealing with both hardware and software). Low level sound and midi drivers reviewed will include oss, oss-free, alsa. Environments for sound synthesis and composition will include the Common Lisp based clm system, STK (c++), and pd (c). Many other interesting tools like the snd sound editor (and its internal scheme programming environment) will also be covered. Due to the very dynamic nature of the open source community and software base more programs will probably be included by the time the workshop starts. The workshop will also include a brief tour of sound processing and synthesis techniques. Familiarity with computers and programming languages is helpful.

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

  This course will cover analysis and synthesis of sounds based on spectral and physical models. Models and methods for synthesizing real-world sounds as well as musical sounds will be presented. The course will be organized into morning lectures covering theoretical aspects of the models, and afternoon labs. The morning lectures will present topics such as Fourier theory, spectrum analysis, the phase vocoder, digital waveguides, digital filter theory, pitch detection, linear predictive coding (LPC), high-level feature extraction, and various other aspects of signal processing of interest in sound applications.

  The afternoon labs will be hands-on sessions using SMS and the Synthesis ToolKit in C++ , and other software systems and utilities. Familiarity with engineering, mathematics, physics, and programming is a plus, but the lectures and labs will be geared to a musical audience with basic experience in math and science. Most of the programs used in the workshop will be available to take home.

  Given the short duration of the workshop and the broad spectrum of topics to cover, the lectures will necessarily be fairly high level in nature. However, a full complement of in-depth readings will be provided for those who wish to investigate the details of the material. Also, the last two days of the workshop will include a more detailed treatment of some advanced topics and the corresponding afternoon labs will give the students a chance to solve some specific problems of their interest.

- **Physical Interaction Design for Music I & II**

  These workshops integrate programming, electronics, interaction design, audio, and interactive music. Focus will be on hands-on applications using sensors and microprocessors in conjunction with real-time DSP to make music. Specific technologies will include Parallax's BasicStamp microprocessor (see `http://www.parallax.com`), PD and/or Max/MSP. Participants will design and build working prototypes using a kit that can be taken home at the end of the workshop.

  These workshops will consist of half-day supervised lab sessions, and half-day lectures, classroom exercises and discussions. Classroom sessions will feature live demos and/or concerts of interactive music and instruments. Participants are encouraged (but by no means required) to bring their own laptop computers with any music software/hardware they already use. Ideally, participants will enroll in both one-week workshops, though this is not necessary.

5 Compositional Activities

5.1 Overview

Since the late 1960's, much of the compositional work at CCRMA has involved a software environment which evolved from the Music V program, which was originally developed at Bell Labs by Max Mathews and his research group. The hardware and software has changed and improved greatly over the decades. 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 (Intel based PCs, SGI's, and NeXTs) running Linux, Irix, and NEXTSTEP operating systems. The functionality of PLA exists now in the form of Common Music (CM) (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 run in real time on fast workstations.

Continuity has been maintained over the entire era. For example, scores created on the PDP10 or Samson Box have been recomputed in the Linux and NeXTStep computing environments, taking advantage of their 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 . Other computers and software are also used for composition. Several composers have realized pieces which make extensive use of MIDI equipment. Readily available commercial software for manipulation of digital audio has brought renewed interest in real-time control and computer-based "musique concrète." The programming environments being used for composition and developmental research include MAX, Patchwork, Smalltalk, Common Lisp, STK, C/C++, and Pd.

Since its beginning, works composed at CCRMA have been highlighted at music festivals, concerts and competitions around the world. Compositions realized at CCRMA have been performed at nearly every International Computer Music Conference; 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 throughout the world. Compositions from CCRMA have also won major electroacoustic music prizes over the 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, Allegro, and others. 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, 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:

Oded Ben-Tal

- *A cote the son cher corps endormi* (2002) for baritone, 4-channel tape, and electronics. Setting of text by Artur Rimbaud.
- *String Quartet* (2001)

  The three movements of this piece were conceived as windows into a larger, single movement quartet taken, as it were, from the beginning, the middle and towards the end of that imaginary quartet. In the first movement each player presents a different musical idea with little or no regard to the others. In the second and third movements these materials begin to interact and intermingle. Thus the tension between the player and the group, which I believe is inherent to the string quartet, provides the conceptual foundation for this piece.
- *Precipitating Lights* (2001) Concerto for percussion and wind ensemble, written for percussionist Michelle Daniels.
- *fortepiano* (2000) for 4-channel tape utilizing the piano as a sound source.
- *How silent comes the water* (2000) for solo piano. Commisioned by the pianist Michal Tal.
- *Saraband* (2000) for flute, clarinet, violin, cello, piano, percussion, and computer generated sound.

Jonathan Berger

- *My Lai* (2001) for solo piano, premiered at the United Nations General Assembly, January 24 2001.
- *Echoes of Light and Time* (2000)

  A sound installation and collaboration with Dale Chihuly. Commissioned by the City of Jerusalem and Chihuly Studio for *Chihuly in the Light of Jerusalem*, 2000. *Echoes of Light and Time* was heard by over two million visitors and received international attention. The CD adaptation of the work was released by Vitosha/Sony in January 2001.
- *Marcatto Sempre* (2000) for clarinet (commissioned by Ensemble Kaprizma).

Christopher Burns

- *The Location of Six Geometric Figures* (2001) for flute, clarinet, violin, cello, piano, and percussion.

  The Location of Six Geometric Figures takes its title from a work by Sol LeWitt, in which areas are compulsively measured and subdivided in order to "locate" simple geometric forms like squares and triangles. The innocuous result is the product of a labyrinth of obsessively documented calculation; the trappings of order cannot disguise the ultimately irrational nature of the project.

  I was inspired by the way LeWitt works out simple schemes to an exhaustive end, and then lets the systems interfere with one another to produce complex results. My work combines several such mutually interfering grids. The durations and subdivisions of the six large sections of the piece, accelerated dramatically, were used to generate the rhythmic details of the work. An independently derived series of bar-lengths, with its own set of repetitions and variations interrupts the flow of time. Instrumental combinations (a solo playing against two independent duos, a single trio) were permuted exhaustively, then re-ordered from "playing independently" to "playing together." Behind these overdetermined systems, the abyss is waiting.
- *Maxwell's Demon* (2001) for five improvisers (open instrumentation) with conductor.

  Consider the oxymoron of composing for improvisers: I wanted to create a piece as open as possible, so that a talented group of improvisers would be given the opportunity to exercise their skills. At the same time, I wanted to compose: to write a piece with a fixed identity, something recognizable from performance to performance. I hit upon the idea of composing only the durations of the piece. Entrances, exits, and the lengths of phrases are all specified, so precisely that a conductor is necessary. But that's all the score contains - the rest is up to the improvisers, who are more than equal partners in making the music. Maxwell's Demon is dedicated to Matt Ingalls.

- *Xerox Book* (2001) Nine miniatures for piano and percussion.

  Xerox Book, for piano and percussion, is a pendant to my sextet *The Location of Six Geometric Figures*. In several movements of the duo, extracts from the larger work are molded and twisted through a variety of idiosyncratic transcription techniques. In other movements of *Xerox Book*, newly composed materials were subjected to similar processes of compression and distortion. In most of the larger movements, there were several generations of transformation before the music reached its final state - just as a sequence of photocopying will gradually distort an image into something new and unrecognizable.

- *Misprision* (2001) for flute and guitar.

- *Gineman* (2000) for harpsichord.

  In Bali, a "gineman" is the introductory section of a longer work for gamelan gong kebyar, characterized by fast, angular phrases. While *Gineman* is certainly not a piece of Balinese music, it borrows the stop-and-start rhythms and unexpected outbursts of its namesake. My perceptions of Balinese modality and rhythmic cycle are also important to the piece, although in a more abstract fashion.

  The harpsichord, closely associated with the European Baroque, may seem an unusual vehicle for music inspired by Bali. However, the two manuals and flexible tuning of the instrument enabled me to approximate the paired tunings that characterize the shimmering soundworld of the gamelan. *Gineman* is dedicated to Mary Francis.

- *Fabrication* (2000) for trumpet and electronics.

  Fabrication begins with a series of fragments: isolated elements of trumpet technique, like breathing and tonguing, are presented divorced from ordinary playing. The acoustic study of the trumpet continues with other splinters of material. Natural harmonics are used to produce distortions of pitch and timbre, and the performer creates further acoustic disruptions with mutes, and by singing into the instrument while playing. Eventually a more normal trumpet technique emerges from the shards, and a kind of calm is achieved. If the piece begins by metaphorically constructing the trumpet from the components of technique, it ends with a more literal disassembly.

  While *Fabrication* is obsessed with trumpet acoustics, it is entirely dependent upon electronics. Many of the sounds used in the piece are too quiet to be heard in performance. And so the microphone serves as a microscope, revealing otherwise inaudible sounds. The electronics gradually take on an active role as well - transforming and extending the sound of the trumpet beyond its acoustic limits.

- *78* (2000) for clarinet, violin, and piano.

  In the 1920s and 30s, New Orleans jazz traveled the world. One of the places where it touched down was Batavia, a region on the outskirts of Jakarta, the capital of Dutch colonial Indonesia. Local jazz bands performed across the region, while 78 records like the Louis Armstrong "Hot Five" and "Hot Seven" sides were broadcast on the radio. The 78 made global musical transmission possible to an extraordinary extent.

  Today, the tanjidor and gambang kromong musics of Batavia present a striking fusion: New Orleans jazz played on traditional Indonesian and Chinese instruments. Or is it jazz musicians trying to reproduce the sounds of Javanese and Sundanese gamelans? It's difficult to say.

  78 continues this cycle of hybridization, bringing elements from tanjidor into my own musical language in a piece which would fit on the two sides of a 78 record. The tightly woven counterpoint, multiple melodic idioms, and structural cycles I've borrowed from tanjidor are recreated here in very different form. But think of the ensemble as a jazz clarinet, a Chinese fiddle, and a set of tuned percussion, and you'll begin to get the idea.

C. Matthew Burtner

- *Ukiuq Tulugaq (the Winter Raven)* (1998-2002)

  Ukiuq Tulugaq is a large-scale multimedia work for voice, instrumental ensemble, electronics, dance ensemble, video projection and theatrical staging. The piece is based on ecological and anthropological studies of the Arctic. The composition metaphorically connects an Inuit creation story, in which the World is created by Raven from snow, with the ecological seasonal approach of Winter. The dramatic shape of *Ukiuq Tulugaq* is a gradual change from Fall into Winter. Each of the three acts explores a different psychological state based on the juxtaposition of time in relation to the ecological change.

  Act 1 takes place before Winter. It is Fall. The family is preparing wood, leaves are falling, there is abundant sunlight. Act 2 is the transformation into Winter. A stark geographic environment created in Kunikluk, is the context for a juxtaposition of the spirit/flesh and the industry/flesh in the Speaking Flesh and Industrial Garden movements. The implied drama of the act is broken by the coming of Ice and Winter. Act 2 moves away from the emotional state occupied by human drama towards a type of environmental drama.

  Act 3 explores this ecological drama. It snows in this act and Raven appears. The wind blows and leaves impressions on the snow. The light darkens and shadows leave impressions on the snow. The animals seek shelter and their fading prints leave impressions on the snow. While the act is predominantly concerned with the movement into a still place of Winter, it also revolves around memory and cyclical processes. The focus on wind itself is a memory of Act 1, and Raven appears, invoking memory (through the use of radio transmitters). This final act concerns the continuation and cohabitation of humanity and nature.

- *Polyrhythmicana* (2002) for flute, cello, guitar, percussion and 4-channel computer-generated click track. Commissioned by San Diego New Music for Ensemble Noise.

  The form of *Polyrhythmicana* is generated by macro-level geometric rhythmic relationships arising from the interplay between the individual instrumental lines. In order for the performers to follow the constantly changing tempi (which are both independent from and closely related to one another) a computer-program was created that generates independent multichannel click tracks under one global clock. The piece is in five movements each with a different rhythmic organization: I: Metal YX; II: Split/Joined Diamonds (in Wood); III: C Acceleration Phase; IV: Slow 2:3 (in Noise); V: Melody Triangles.

- *Animus/Anima* (2001) for voice and electronics.

  Commissioned by Haleh Abghari. First performed at Stanford.

- *S-Trance-S* (2001) for computer metasaxophone.

  For Stefania Serafin, this piece explores new expressive possibilities arising from instrument controller substitution (Burtner, Serafin 2001). First performed at Mills.

- *Delta* (2001) for electric saxophone.

  The electric saxophone is an ongoing part of the metasaxophone project involving microphones imbedded inside the saxophone, distortion, and chaotic feedback systems. First performed at CCRMA and CNMAT with Earsay.

- *Natigviksuk* (2000) for viola, alto saxophone, piano, and noise generators.

- *Studies for Radio Transmitter* (2000) for home-made radio transmitters.

- *Oceans of Color* (2000) for 27 solo saxophones.

- *Signal Ruins* (2000) for prepared piano, bass drum, and electronics.

Chris Chafe

- *Double* (2002) for clavichord and CD.

  "Oxygen Flute" music installation, 2001, San Jose Museum of Art.

- *Ping* (2001)

  Created by composer and researcher Chris Chafe and digital artist Greg Niemeyer, *Ping* is a site-specific sound installation that is an outgrowth of audio networking research at Stanford University's Center for Computer Research in Music and Acoustics and interactive and graphic design experiments originating from the Stanford University Digital Art Center. *Ping* is a sonic adaptation of a network tool commonly used for timing data transmission over the Internet. As installed in the outdoor atrium of SFMOMA, Ping functions as a sonar-like detector whose echoes sound out the paths traversed by data flowing on the Internet. At any given moment, several sites are concurrently active, and the tones that are heard in Ping make audible the time lag that occurs while moving information from one site to another between networked computers.

  Within the *Ping* environment, one can navigate through the network soundscape while overlooking San Francisco, a cityscape itself linked by the same networks that constitute the medium. Visitors to the installation can expand or change the list of available sites as well as influence the types of sound produced, choosing different projections of the instruments, musical scales, and speaker configurations in the surround-sound environment.

  Current explorations pertaining to sound synthesis and Internet engineering are the foundation of the Ping installation. The research that led to this installation is, however, just one part of a larger effort to investigate the usefulness of audio for internetworking and, reciprocally, ways in which the Internet can abet audio.

Ching-Wen Chao

- *Elegy in Flight* (2002) for violin.

  Elegy in Flight was recently premiered by Mark Menzies on January 22, 2002.

- *Departure Tracings* (2000) for sextet.

  Departure Tracings is the second in a series of works dedicated to the memory of my father. Each work in the series utilizes the pitches C and G# as points of departure and/or arrival (these two pitches come from my father's initials). *Departure Tracings* was premiered by EARPLAY on May 1, 2000. The Cal Ear Unit Ensemble gave it a wonderful presentation in October 2000 and February 2001.

- *The Captured Shadow* (2001) for soprano trombone, delay, and pre-recorded tape.

  The Captured Shadow pursues a theatrical aspect of live electronic music. Inspired by novels of Fitzgerald's, the piece experiments with the representation of literal meanings in music, such as "betrayal" and "emptiness." The work utilizes speech-like materials and the pitch flexibility of the soprano trombone to present a vague story-telling voice. This narrator, though often obscure, creates a context for the musical representation of literary ideas. I am indebted to Chris Burns for his help in every aspect of this work.

- *SOUNDSTATES* (1998/2001) for percussion and tape.

  Soundstates explores the 3 states of matter (gas, liquid and solid) and their transformations into one another. This flowing from one sound state to the other forms the basis of the structure of the piece, to reflect a similar process in the spontaneous changes in nature. The piece begins with solid, block-like sounds which gradually disintegrate; it ends with a succession of rising, more atmospheric sounds, with a return to elements of the original material. The source sounds were mostly drawn from the marimba and were digitally processed in the CLM (Common Lisp Music) environment. Many thanks to Juan Pampin who helped me in employing CLM instruments.

- *Elegy* (2000) for celletto.

  Elegy is the third in a series of works dedicated to the memory of my father. Each work in the series utilizes the pitches C and G-sharp as points of departure and/or arrival (these two pitches come from my father's initials).

  This piece for Chris Chafe's special instrument, the celetto, is concerned with the purification of tone. C and G-sharp are highlighted, but they are treated as anchors in a larger pitch world that expands around them. Elegy could be viewed as a complement to my third study for two pianos, which is another work in this series.

  Elegy was premiered by Chris Chafe with his celleto in the CCRMA-CNMAT exchange concert in April 2000, and recently presented in the Seoul Electronic Music Festival in Korea in Nov 2000.

Damián Keller

- *Paititi: a multimodal journey to El Dorado*, an installation.

  Paititi is a multidisciplinary project encompassing historical research, software development, and artistic creation. The main objective of this project is the elaboration of a work involving literary, visual and sonic elements inspired and documented on the historical records of the legend of El Dorado.

  The temporal frame for this piece is the Spanish-Inca war (1532-1572). The geographical region includes the paths taken by the first Spanish explorers and some of the important Inca cities, such as Cuzco, Quito and Cajamarca. The visual material includes reproductions of historical documents from the Archivo General de Indias (AGI) in Seville and original footage taken at the sites of the explorations. The sonic material features recordings on site and oral reports by people from the region. The literary material emcompasses documents written by the conquistadores, oral reports by aboriginals, and original prose inspired on historical facts.

  The composition will be created by means of digital processing and synthesis of environmental sounds modelled after the collected material. The format of the piece will be an installation space combining video, still digital images, and multichannel sound. Two types of software will be developed for this work: sound synthesis and multichannel spacialization, and interactive controllers for triggering images and sounds in the installation space.

- *Instábilis* (2001). Installation with Karina Yaluk.

  Instábilis represents a extreme in the spectrum of compositional techniques based on the ecological perspective. This piece proposes a finite sonic space delimited by the resonant characteristics of a single body: a metalic sculpture. As an analogy to the visual objects exposed in the installation, the sculpture is explored from different angles creating the timbre space of the piece.

  The temporal structure - camparable to a kaleidoscope - is ramdom, modular, and constantly varying. Two stereo sources with similar sonic material reflect the symmetry of the space. Nevertheless, as it is the case in environmental sound, the sonic events - with a total duration of twenty minutes - contain no literal repetitions. Given this aleatoric structure, each listener experiences a unique version of the piece.

  During 2001, *Instábilis* has been presented at Espacio Cultural Citibank, Asunción, Paraguay, and Galeria Athos Vulcão, Brasília, Brazil. This piece was realized at the studios of Universidade de Brasília.

- *Dorotéia* (2001), theater soundtrack with Ana Lúcia, F.S. Keller, and Guilherme Coelho.

  The sound material for *Dorotéia* was organized using three sound classes which served as perceptual axes for structuring the piece: vocal sounds, water drops and hybrid metal, water and vocal events. Sonic transformations consisted in temporal and spectral transitions between these classes.

  The combination of a granular control data structure with constrained sonic databases allowed us to synthesize hybrid events, featuring characteristics of different sound classes. The extraction of events from ambient recordings provided a way to relate the sources with their placement in the space. The spatial layout consisted of three separate stereo tracks distributed in the performance space. Outdoor recordings were utilized for open spaces and reverberation by convolution was employed for enclosed spaces.

During the performance, the sound track consisting of water drops was played continuously, the vocal track established a dialogue with the actress, and the complex events were played at key moments in the piece.

This work was funded by the Brazilian Student Association of Stanford University. Presented on May 3, 4, and 5, 2001, during the Brazilian Week at the Elliot Center, Stanford University, CA.

- *La Conquista* (2001), an installation with Ariadna Capasso.

  La Conquista addresses the theme of the Spanish conquest and the dynamics of power relationships between aboriginals and conquerors. This piece was based on the soundtrack 'Oro por Baratijas' [Compact Disc Organised Sound 5(3)].

  The sonic composition provided a template for the video editing process. This was done on a Dell computer, using Premiere 5.0. The raw footage was recorded with a Sony camera on Digital 8 format. The four natural elements, earth, water, fire and air, were used as basic sources to create leitmotifs that reflect the forces of nature and the dynamics of social interactions.

  La Conquista has been presented in Boulder, Denver, Boston, New York, Buenos Aires and Havana.

- IQ^2 (2000), an interactive installation with Una Knox.

  Current capitalist society functions on the basis of circulation and accumulation of goods. IQ^2 is also shaped after these two processes. The number and behavior of people in the space define the number and characteristics of the events being triggered. Thus, IQ^2 could be compared to an organism that reacts to human presence. Because the triggering process is based upon the sensors' change of state, IQ^2 is not excited by too many stimuli. In fact, if all sensors are constantly active, no event is triggered. IQ^2's "quiet" state follows an eighty-hour cycle. As long as it is not disturbed, the piece will repeat itself every three days and eight hours.

  The material in IQ^2 consists of a few sonic grains and video images. Thus, the local elements of the piece are very static, almost immobile. As it is the case in human social systems, the most relevant structures take place at a higher level. They are determined by relationships among a large number of elements and by processes which unfold very slowly.

  IQ^2 takes shape at the junction between time and space. People's behaviors in IQ^2's space establish what events occur and how they are distributed through time.

  IQ^2 was presented in September 2000 at the Industrial Ear Exhibition, in the Ironworks Gallery, Vancouver, Canada. This work was funded by and realized at the Western Front Artist-Run Center, Vancouver.

- *Metrophonie* (2000) for four-channel tape.

  Metrophonie, as its title suggests, is loosely inspired on Stockhausen's Mikrophonie. The similarities are: the title, the four-channel format, the transformation of recorded sounds and the subdivision into sections, in other words, all that doesn't matter.

  Metrophonie takes place in San Francisco's metro, (aka BART). A woman pays a fare; a drunkard sings; a big man hugs his buddy; and a "mama works at Brian's tonight." As in San Francisco's street life, money is very important here. But more important is the lack of money.

  Some streets in San Francisco are not clean. Some people in these streets are also not very clean. The sounds these people make are rich. That's why it doesn't matter whether they are dirty or clean, whether they are black or white. By the way, what time is the next blackout?

- *The Urban Corridor* (2000), an interactive installation with Ariadna Capasso and Patricia Tinajero-Baker.

  The *The Urban Corridor* was presented from June to August 2000 at the CU Art Galleries, Boulder, within the context of the Electronic Easel exhibition. The installation consisted of a constructed space shaped as a corridor containing lights, motion sensors, two slide projectors, a video projector, and a multichannel sound system. The whole setup was run from a Macintosh PPC computer equipped with two CD-ROMs and an x10 two-way interface.

  This installation is a collage of a video projection with sound compositions. The sounds elicit images of the urban environment. There is a soundtrack for the video and several sets of independent sounds which are triggered by the visitors. These sounds are connected to a computer with two CD-ROMS that are randomly activated. There is another set of environmental sounds coming from a CD player which constantly play. The video interweaves four 'news' events: an automobile accident, a public demonstration, urban sprawl, and a war. These events are designed to encompass the audience perception from a global (i.e. a war) to an individual experience (i.e. car wreck). The news expose the process by which real occurrences become a mediated reality. The video utilizes a combination of found, original footage and voice-overs. We create a virtual cityscape with a large video projection and surround sound which envelope the viewer. This is an interactive piece. As the participant enters the room, he/she triggers the video projection and audio sound. As the participant walks around the room, he/she turns on motion detectors which trigger the sounds.

- *P2000* for compact bassoon, disc, glass, and eight-channel tape. Commissioned by Uruguayan bassoonist E. Donas Goldstein.

- *The Trade*, soundtrack for installation. Collaboration with visual artist Ariadna Capasso, Boulder, CO.

- *Working Title*, sound installation. Commissioned by SoundCulture 2000, Vancouver.

Seungyon-Seny Lee

- *Idiosyncrasy (Pleasure, Anger, Lament and Joie)* (2001), a real-time interaction composition for a mezzo-soprano & video.

  In a sense of color, how much blue can be made the tone of dark-grass-green with yellow, and how much of the same blue can be made yellowish green with the same yellow? What makes it possible to think in certain ways, and to not think in other ways?

  Psychoanalytic issues drove the composer to portray the individual mirror image of Self and Other through the fundamental emotions of human being, which include at least these four: Pleasure, Anger, Lament, and Joie (Hee-Ro-Ae-Rok) in the piece, *Idiosyncrasy.* The composition draws on three poems in three different languages, both using the essential meaning of the words and liberating their phonetics from the lexical hindrances of a given time and place. Many thanks for Cyrille Brissot who collaborates for video images, and also for Takayuki Nakano who lets me use a part of his poem.

- *Je est un Autre (Self is Other)*

  Je est un Autre is a journey of the imagination and an aesthetic evolution of its ingredients. The cycle consists of four pieces, each composed for different media. Electro-acoustic music and visual images are combined in the first piece, dance, video images, installations, and computer generated sound in the second, instrumental sextet and theater in the third, and a sound installation with ceramics in the fourth.

  The raw acoustic materials were recorded of the sounds from nature, such as running water, shaking rice, insects, etc. These sounds were electronically processed, creating a broad palette of timbres and sonic textures. The sound transformations in the cycle, *Je est un Autre* are used to develop diverse musical layers, inviting the listener to interpret the music through his own imagination.

Throughout the pieces, imagination, as a metaphor for the unconscious, signifies the existence, which struggles within an ego in memory and in reality, from nil to eternity. The three stages - the Real (need), the Imaginary (demand), and the Symbolic (desire) - that I represent in the pieces come from a notion of Jacques Lacan.

- *Je est un Autre I* (1999), tape and installation.

  In *Je est un Autre I*, a fish tank is placed in front of the video projector. The shadow effect of rippling water delivers images, which refer to the unconscious as the foundation of all being. Imagery for the piece is included animation and photograph images, which were chosen for their symbolic reference to internal psychological states.

- *Je est un Autre II* (2001), for tape, 3 dancers, video images, and installations.

  Dance in *Je est un Autre II* is used continue the exploration of concepts originally presented *Je est un Autre I.* The three dancers roughly signify the three phases of becoming an individuated being, as theorized by Jacque Lacan; the Real (need), the Imaginary (demand), and the Symbolic (desire). Choreography was influenced by 'ABECEDA' by Karel Teige and Vtezslav Nezval (1926). The gestures of the dancers depict letters of the alphabet, spelling out terminology used by Lacan in French.

  The installation is intended to convey aspects of Lacan's linguistic structure of discourse. The three panels used as props by the dancers represent the phases of the linguistic process. Discourse originates in the unconscious (represented by the plastic sheet). The abstract form of the idea filters through the memory (the transparent scrim), and is formulated as language (the newspaper sheet). The imaginary phase of the unconscious is further represented in the piece by projected images. The photographs of the abstract images were chosen for their symbolic references to internal psychological states.

  Thanks to Juan Pampin for the use of his software, ATS (spectral Analysis, Transformation, and Synthesis of sounds) and Kotoka Suzuki for editing the video.

Fernando Lopez Lezcano

- *iICEsCcRrEeAaMm*

  iICEsCcRrEeAaMm is a beta, er.. I mean alpha version of a new multichannel tape piece I'm still working on. As in the software world, Marketing informs me that in future versions bugs will be squashed and new features will be added for the benefit of all listeners. iscream refers to the origin of most of the concrete sound materials used in the piece. Screams and various other utterances from all of Chris Chafe's kids were digitally recorded in all their chilling and quite upsetting beauty. They were latter digitally fed into the "grani" sample grinder, a granular synthesis instrument developed by the composer. ICECREAM refers to the reward the kids (and myself) got after the screaming studio session. The piece was composed in the digital domain using Bill Schottstaedt's Common Lisp Music. Many software instruments and quite a few other samples of real world sounds made their way into the bitstream.

Charles Nichols

- *Strata 2* for flute and interactive computer programming (2001)

  Strata 2 is a study in obscuring and defining harmonic motion, obstructing and establishing rhythmic pulse, animating surface detail, and signal processing with modulation techniques.

  The piece is divided into four sections, with an additional introduction and two brief interludes. Each section is further divided into seven subsections, each of which are based on one of three harmonies, eight- and nine-pitch groups, which extend through the range of the flute. The four sections move from obscured to defined harmonic motion, through the use of greater or fewer auxiliary pitches, which revolve around the primary pitches of the harmonies.

  These sections also move from obstructed to established rhythmic pulse, through the use of greater or fewer rhythmic interruptions and grace notes, and expansion and contraction of sustained notes.

The sustained notes are animated with trills and vibratos of three different speeds, flutter tongues, and sung pitches, which create interference with the timbre of the flute. The timbre of the flute is further processed with computer programming, using amplitude- and ring-modulation, and spatialized around four speakers.

Jonathan Norton

- *Three Sides of a Coin* (2002) for flute, clarinet, percussion, piano, violin and cello.
- *Newspaper Music: German Edition* (2002) for solo piano. Part of a series of MAX/MSP algorithmically composed compositions based on newspaper articles from around the world.
- *Stop Thief!* (2002) for tape.

  This text-sound music piece is based on a text about the return of a "new legitimate" Napster as a subscription music service.
- *String Quartet* (2001) for two violins, viola and cello.
- *Meet Your New Neighbors* (2001), the soundtrack for a documentary chronicaling the stories of three people, from Africa, Vietnam, and Bosnia, who triumph over adversity and arrive in America.
- *Der Friedensstifter (Peacemaker)* (2001) for oboe, piano, percussion and cello.

Juan Reyes

- *Chryseis* (2002) for 4-channel tape.

  Scattering of names like Achilles, Braiseis or Chryseis can only come from the old world. The probability of selecting such a name while in Ibero America might well be very odd. This is like choosing characters for a play or a novel, a login name for haut mail, a password or perhaps a new weather phenomenon in the Caribbean. Nevertheless, this name sounds like two syllables barely pitched if whispered but very flexible if shout. Achilles, Braiseis or Chryseis are expressive while sung in bossa nova or at la cosa nostra.

  This is yet another composition for systems which mimic the vibrational properties of a musical sound. In this case Scanned Synthesis developed by Bill Verplank and Max Mathews at CCRMA during the last years of the past century, was used as the underlying material. The process for achieving this timbre was solved by scanning and manipulating several types of springs which give different and time mutant spectra. Control is achieved by mathematical modeling the haptics of the spring.

  Scanned Synthesis is based on the psychoacoustics of how we hear and appreciate timbres and on our motor (haptic) abilities to manipulate timbres during performance. It involves a slow dynamic system whose frequencies of vibration are below 15Hz. The system is directly manipulated by motions of the performer. The vibrations of the system are a function of the initial conditions, the forces applied by the performer and the dynamics of the system.

  This piece was composed using the Common Lisp Music and Common Music environments on Linux at CCRMA.
- *Wadi Musa or The Monteria Hat* (2001) for quenas (Andean pan flutes), cello, and physical models of maracas and clarinet.

  Wadi Musa or the Valley of Moses was the city of the Nabateans some centuries ago. A "rose-red city half as old as time," where sand has witnessed unconscious listeners of the whisper of dessert creatures, wind, and water. Well deep there, two thousand and one steps below, there is the Monteria Hat, a curious object indeed.

  This is a composition for quenas (Andean pan flutes), cello and physical models of maracas and clarinet. The maracas belong to special breed of models developed by Perry Cook called PHISM (Physically Inspired Sonic Modeling). The polyrhythms are the result of combinations somewhat chaotic between the shake rate, seed quantity, and size of shell, thus inspired by the Monteria Hat.

- *Oranged (lima-limon)* (2001) for tape

  Oranged (lima-limon) are colors with bright spectrum out liners of geometric segments and shapes over gray scale pictures. In this music they synthesize several combinations contrasting over a variety of shades of noise...

  Oranged (lima-limon) is also a fragrance and personality: An orange jacket on a bright day and a yellowish lime jacket at night. The spirit of freshness and love for dogs.

  This piece was composed using Frequency Modulation and only Common Lisp Music on Linux at CCRMA.

- *ppP* (2000) for piano and tape

  ppP in its concert version (there is a museum version), is an algorithmic composition for traditional acoustic piano and modeling of the piano. This piece uses a computer model of a piano in an unusual tunning as contrast and complement to the real instrument on-stage. The software piano has indefinitely vibrating strings, non-standard temperaments and different string lengths and densities for the same pitch. In this piece the physical model has been tuned to the Bohlen Pierce scale. Additionally, the context surrounding the string can change-it need not be struck by a hammer or resonated sound-board. ppP, stands for perfectly pitched piano or perfectly perceived piano but also might also mean pianissimo and rather not in regards to dynamics. This piece was composed using Scott Van Duyne's Physical Model of the Piano developed at CCRMA in Common Lisp Music.

Kotoka Suzuki

- *Sift* (2000) for violin and computer-generated tape

  Sift was commissioned by MATA and is dedicated to Carla Kihlstedt. This piece conveys the relationship between two elements of sound: noise and pitch. These two elements are emphasized as separate voices by assigning each to an instrument: noise to tape and pitch to violin. Throughout the piece, the exchange of these elements, and the transformation from one element to another can be heard. All computer-generated sounds are derived from the sounds of the violin used in this work. Similarly, the violin often imitates the sound of the computer-generated material on the tape. The violin sounds were manipulated and recorded for the music of the tape using sound editing programs such as CLM, Snd, and Pro Tools.

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 *grani*, a granular synthesis instrument for CLM

Fernando Lopez Lezcano

grani.ins is a quite complete CLM (Common Lisp Music) granular synthesis instrument designed to process (ie: mangle) input soundfiles. Almost all parameters of the granulation process can be either constant numbers or envelopes so that a note generated with grani can have very complex behavioral changes over its duration. Parameters can control grain density in grains per second, grain duration, grain envelope (with up to two envelopes and an interpolating function), sampling rate conversion factor in linear or pitch scales, spatial location of grains, number of grains to generate or duration of the note, etc. Almost all the parameters have a companion "spread" parameter that defines a random spread around the central value defined by the base parameter (both can be envelopes).

The first "grani" instrument was originally created as an example instrument for the 1996 Summer Workshop. In its present form it has been used to teach granular synthesis in the 1998 Summer Workshop and 220a (Introduction to Sound Synthesis Course). It has become a pretty popular instrument at CCRMA and was used by its author to compose *iICEsCcRrEeAaMm*, a four channel tape piece that was premiered in the 1998 CCRMA Summer Concert.

Complete details can be found at: `http://www-ccrma.stanford.edu/~nando/clm/grani/`

6.1.2 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. Complete details can be found at http://www-ccrma.stanford.edu/~nando/clm/dlocsig/.

dlocsig.lisp is a unit generator that dynamically moves a sound source in 2d or 3d space and can be used as a replacement for the standard locsig in new or existing CLM instruments (this is a completely rewritten and much improved version of the old dlocsig that I started writing in 1992 while I was working at Keio University in Japan).

The new dlocsig can generate spatial positioning cues for any number of speakers which can be arbitrarily arranged in 2d or 3d space. The number of output channels of the current output stream (usually defined by the :channels keyword in the enclosing with-sound) will determine which speaker arrangement is used. In pieces which can be recompiled from scratch this feature allows the composer to easily create several renditions of the same piece, each one optimized for a particular number, spatial configuration of speakers and rendering technique.

dlocsig can render the output soundfile with different techniques. The default is to use amplitude panning between adyacent speakers (between two speakers in 2d space or three speaker groups in 3d space). dlocsig can also create an Ambisonics encoded four channel output soundfile suitable for feeding into an appropriate decoder for multiple speaker reproduction. Or it can decode the Ambisonics encoded information to an arbitrary number of output channels if the speaker configuration is known in advance. In the near future dlocsig will also be able to render to stereo soundfiles with hrtf generated cues for heaphone or speaker listening environments. In all cases doppler shift is also generated as well as amplitude scaling due to distance with user-defined exponents and ratio of direct to reverberated sound.

The movement of sound sources is described through paths. These are CLOS (Common Lisp Object System) objects that hold the information needed by dlocsig to move the source in space and are independent of the unit generator itself. Paths can be reused across many calls to dlocsig and can be translated, scaled and rotated in space as needed. There are several ways to describe a path in space. Bezier paths are described by a set of discrete points in 2d or 3d space that are latter joined by smoothly curved bezier segments. This description is very compact and easy to specify as a few points can describe a complex trajectory in 3d space. Paths can also be specified in geometric terms and one such implementation (spirals) is currently provided.

The dlocsig unit generator uses the same interface as all other CLM unit generators. make-dlocsig creates a structure for a given path and returns (as multiple values) the structure and the beginning and ending samples of the note. dlocsig is the macro that gets compiled inside the run loop and localizes the samples in space.

6.1.3 Planet CCRMA at Home

Fernando Lopez-Lezcano

Planet CCRMA at Home is an ever-growing rpm package collection that includes almost all the extra software that is installed at CCRMA on top of a normal RedHat Linux operating system install. RPM, or RedHat Package Manager, is a set of programs that make it easy to install, remove, and update software packages in an rpm-based linux system (rpm is used by RedHat, Mandrake and SuSE amongst others). The latest version is managed through `apt` for `rpm`, a tool for dependency control that enables easy installation of packages, with automatic dependency tracking and retrieval from a network accessible repository. Packages include a recent linux kernel with low latency patches installed, a current version of the ALSA sound driver packages, a wealth of music, sound, midi and office applications and various updates and additions to some packages that are part of the standard RedHat install.

The goal of *Planet CCRMA at Home* is to provide an easy way to "upgrade" a RedHat install to include most (if not all) of the custom software that makes up the Linux based sound and music computing environment at CCRMA.

For more information, see `http://www-ccrma.stanford.edu/planetccrma/software/`.

6.1.4 Planet CCRMA

Juan Reyes and Fernando Lopez-Lezcano

Planet CCRMA is an HTML document for the purpose of illustrating and informing new CCRMA users and visitors about the computer resources, the Linux environment, and applications which might be helpful for doing research and compositional work at CCRMA. It also briefly describes the meaning of "open source" as a part of a laboratory and community philosophy at CCRMA. It is also a brief history of hardware at CCRMA and descriptions of Linux as an operating system, the Unix environment, useful shell commands and many X windows applications in addition to Gnome and KDE desktops. In the applications section there are descriptions of programs and information provided by the developers' documentation and direct links to the application web page. *Planet CCRMA* focus is as the first stepping stone for a particular command, program or application but nevertheless the reader is encouraged to find more in-depth information on the Unix manual pages, on the web or in the links to home pages which will are also provided. During the 2001 autumn quarter at Stanford the web page was visited by more than 80% of new and old users of the CCRMA network and community. *Planet CCRMA* is also updated on a regular basis as per users suggestions and because of new software, upgrades, or updates to the system.

6.1.5 The Synthesis ToolKit in C++ (STK)

Perry R. Cook and **Gary P. Scavone**

The Synthesis ToolKit in C++ (STK) is a set of audio signal processing and synthesis classes and algorithms written in C++. You can use these classes to create programs that make sounds with a variety of synthesis techniques. This is not a terribly novel concept, except that the Synthesis ToolKit is extremely portable (it's mostly platform-independent C and C++ code), and it's completely user-extensible (no libraries, no hidden drivers, and all source code is included). We like to think that this increases the chances that our programs will still work in another 5-10 years. In fact, the ToolKit has been working continuously for nearly 8 years now. STK currently runs with "realtime" support (audio and MIDI) on SGI (Irix), Linux, and Windows computer platforms. Generic, non-realtime support has been tested under NeXTStep, Sun, and other platforms and should work with any standard C++ compiler.

The Synthesis ToolKit is free for non-commercial use. The only parts of the Synthesis ToolKit that are platform-dependent concern real-time audio and MIDI input and output, and that is taken care of with a few special classes. The interface for MIDI input and the simple Tcl/Tk graphical user interfaces (GUIs) provided is the same, so it's easy to experiment in real time using either the GUIs or MIDI. The Synthesis ToolKit can generate simultaneous SND (AU), WAV, AIFF, and MAT-file output soundfile formats (as well as realtime sound output), so you can view your results using one of a large variety of sound/signal analysis tools already available (e.g. Snd, Cool Edit, Matlab).

The Synthesis Toolkit is not one particular program. Rather, it is a set of C++ classes that you can use to create your own programs. A few example applications are provided to demonstrate some of the ways to use the classes. If you have specific needs, you will probably have to either modify the example programs or write a new program altogether. Further, the example programs don't have a fancy GUI wrapper. If you feel the need to have a "drag and drop" graphical patching GUI, you probably don't want to use the ToolKit. Spending hundreds of hours making platform-dependent graphics code would go against one of the fundamental design goals of the ToolKit - platform independence.

For those instances where a simple GUI with sliders and buttons is helpful, we use Tcl/Tk (which is freely distributed for all the supported ToolKit platforms). A number of Tcl/Tk GUI scripts are distributed with the ToolKit release. For control, the Synthesis Toolkit uses raw MIDI (on supported platforms), and SKINI (Synthesis ToolKit Instrument Network Interface, a MIDI-like text message synthesis control format).

Perry Cook began developing a pre-cursor to the Synthesis ToolKit (also called STK) under NeXTStep at the Center for Computer Research in Music and Acoustics (CCRMA) at Stanford University in the early-1990s. With his move to Princeton University in 1996, he ported everything to C++ on SGI hardware, added real-time capabilities, and greatly expanded the synthesis techniques available. With the help of Bill Putnam, Perry also made a port of STK to Windows95. Gary Scavone began using STK extensively in the summer of 1997 and completed a full port of STK to Linux early in 1998. He finished the fully compatable Windows port (using Direct Sound API) in June 1998. Numerous improvements and extensions have been made since then.

For more information about STK, see `http://www-ccrma.stanford.edu/software/stk/`.

References:

- Cook, P.R. and Scavone, G.P. *The Synthesis ToolKit (STK)*, In Proceedings of the 1999 International Computer Music Conference, Beijing, China, 1999.

- Cook, P. R. *Synthesis ToolKit in C++, Version 1.0*, In SIGGRAPH 1996, Course #17 and 18, Creating and Manipulating Sound to Enhance Computer Graphics, May 1996.

6.1.6 RtAudio: A Cross-Platform C++ Class for Realtime Audio Input/Output

Gary P. Scavone

RtAudio is a cross-platform C++ class for realtime audio input and output streaming. *RtAudio* provides a flexible, easy to use application programming interface (API) which allows complete audio system control, including device capability querying, multiple concurrent streams, blocking and callback functionality. *RtAudio* is currently supported on Windows platforms using the DirectSound API, Linux platforms using both the OSS and ALSA APIs, and on Irix platforms. Support for OS-X and Steinberg ASIO drivers is planned for Spring 2002.

Reference:

- Scavone, G. P. (2002) *RtAudio: A Cross-Platform C++ Class for Realtime Audio Input/Output*, Submitted to the 2002 International Computer Music Conference.

6.1.7 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.

Snd is a sound editor 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; it provides its own music symbol font.

CLM, CMN, and Snd are available free, via anonymous ftp at `ftp://ftp-ccrma.stanford.edu` as pub/Lisp/clm-2.tar.gz, pub/Lisp/cmn.tar.gz, and pub/Lisp/snd-5.tar.gz.

6.1.8 Common Music

Heinrich Taube

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/software/-cm/cm.html` for more information.

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.

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.

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`.

6.2 Physical Modeling

6.2.1 From Physics of Piano Strings to Digital Waveguide

Julien Bensa, **Stefan Bilbao**, **Richard Kronland-Martinet**, and **Julius Smith**

Several models of transverse wave propagation on a piano string, of varying degrees of complexity, have appeared in the literature. These models are always framed in terms of a partial differential equation (PDE), or system of PDEs; usually, the crude starting point for such a model is the one-dimensional wave equation, and the more realistic features, such as dispersion, frequency-dependent loss and nonlinear hammer excitation, are incorporated through several perturbation terms. Chaigne and Askenfelt have proposed the most advanced such model, and used it as the basis for a sound synthesis technique, through the use of finite differences—the time waveform on a struck piano string is simulated in this way to a remarkable degree of fidelity.

Digital waveguides, on the other hand, are filter-like structures which model one-dimensional wave propagation as purely lossless throughout the length of the string, with loss and dispersion summarized in terminating lumped filters. They are thus simulations of slightly modified physical systems, but are highly efficient structures in the context of musical sound synthesis. The aim of this paper is to bridge the gap between PDE models and digital waveguides, and to explicitly show the relationship between the lumped filters used to model loss and dispersion and the parameters which define the model PDE, which, in this case, is a carefully chosen variant of Chaigne and Askenfelt's model system. The calibration of the filters to experimentally measured data is also discussed.

References:

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- J. O. Smith, *Digital Waveguide Modeling of Musical Instruments*, `http://www-ccrma.stanford-.edu/~jos/wg.html`.

6.2.2 Use of Physical Model Synthesis for Developing Experimental Techniques in Ethnomusicology: The Case of the Ouldémé Flute

Patricio de la Cuadra

As part of a collaboration project with ethnomusicologists and the Acoustique Instrumentale Group at IRCAM, we study and implement a physical model of a flue instrument, exploring the flexibility to geometry changes in the instrument, possibilities of timbre control as well as playability of the model. Two implementations have been developed: a C++ object using STK library and an external object for MAX. Implementations have been designed as laboratories, allowing the user to adjust many parameters in real time and evaluate the response of the model. A real-time MIDI controller for a flute physical model is being designed.

The goal here is to study the musical scales of the flutes played by a tribe from North Cameroon: the Ouldémé. In their usage, we encounter vocalic-instrumental polyphony at two levels - individual, where each player plays two flutes, and in groups normally formed by five players. The flutes don't have toneholes and they are made of bamboo cane with a blowing end at one side and a close end at the other. Placing the tongue outside of his mouth, the player shapes the air stream, which then strikes a sharp edge of the cane. The mouthpiece is simply done by cutting the cane and there is no additional work on it. Adding water prevents air leakage and, at a lower degree, adjusts the tuning of the instrument.

These flutes work with a turbulent jet flow. The behavior of this type of jet is less understood than the laminar one and it is currently being studied in a parallel project. A one dimensional representation of the dynamics of the jet (formation, velocity fluctuation, oscillations) is used as described in [1]. The bore is modeled using one dimensional waveguides. Visco-thermic losses and radiation of the sound are implemented as linear filters.

Reference:

1. Verge, M.P. (1995) *Aeroacoustics of confined jets.* PhD thesis, Technical University of Eindhoven, 1995.

6.2.3 Synthesis of a Neolithic Chinese Flute

Patricio de la Cuadra and Chris Chafe

As part of an ongoing project to model the 9000 year old bone flutes unearthed at the Jiahu archeological site in china, we have currently implemented a real-time one-dimensional model of a flute, incorporating previously described techniques for air jet and wind instrument turbulence. The model is implemented in several systems, including STK, and as an external object in Max/MSP and Pd (pure data). It allows for real-time control of fingering, embouchure, and breath pressure.

One application of the research has been as the instrument (a quartet, actually) in a six-month installation at the San Jose Museum of Art entitled "Oxygen Flute." Another application is to recreate the sound of several unplayable archeological specimens with performance practice inferred from traditional playing technique.

6.2.4 Toward High-Quality Singing Synthesis with Varying Sound Qualities

Hui-Ling Lu

To achieve high-quality singing synthesis, spectral modeling and physical modeling have been used in the past. However, spectral models are known to be articulation difficult and expressivity limited. On the other hand, it is not straightforward to adjust physical model parameters to reproduce a specific recording. In this thesis, a high-quality singing synthesizer is proposed with its associated analysis procedure to retrieve the model parameters automatically from the desired voices. Since 95% in singing is voiced sound, the focus of this research is to improve naturalness of the vowel tone quality. In addition, an intuitive parametric model is also developed to control the vocal textures of the synthetic voices ranging from "pressed", to "normal", to "breathy" phonation.

To trade off between complexity of the model and the corresponding analysis procedure, a source-filter type synthesis model is proposed. Based on a simplified human voice production system, the source-filter synthesis model describes human voices as the output of the vocal tract filter excited by a glottal excitation. The vocal tract is modeled as an all-pole filter since only non-nasal voiced sound is focused. To accommodate variations of vocal textures, the glottal excitation model consists of two elements: the derivative glottal wave and the aspiration noise. The derivative glottal wave is modeled by the transformed Liljencrants-Fant (LF) model. Moreover, the aspiration noise is represented as pitch-synchronous, amplitude-modulated Gaussian noise.

The major contribution of this thesis is the development of an analysis procedure that estimates the parameters of the proposed synthesis model to mimic the desired voices. First, a source-filter deconvolution algorithm via the convex optimization technique is proposed to estimate the vocal tract filter from sound recordings. Second, the inverse filtered glottal excitation is decomposed into a smoothed derivative glottal wave and a noise residual component via Wavelet Packet Analysis. Proper parameterizations of the glottal excitation can then be found. By analyzing baritone recordings, a parametric model is constructed for controlling vocal textures in synthesized singing.

References:

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6.2.5 Acoustic Research and Synthesis Models of Woodwind Instruments

Gary P. Scavone

The modeling of musical instruments using digital waveguide methods 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.

Models of wind instrument air columns have reached a high level of development. An accurate and efficient means for modeling woodwind toneholes was described in [Scavone and Cook, 1998]. Another model of the tonehole was developed with Maarten van Walstijn [van Walstijn and Scavone, 2000]. It uses wave digital filter techniques to avoid a delay-free path in the model, thus allowing shorter tonehole heights than is possible with the distributed model of [Scavone and Cook, 1998]. Recently, a study comparing tonehole radiation measurements to digital waveguide and frequency-domain model results was conducted [Scavone and Karjalainen, 2002].

Previous work focused on modeling the direction-dependent sound radiation from woodwind and brass instruments [Scavone, 1999]. The current acoustic theory regarding sound radiation from ducts and holes can be implemented in the digital waveguide context using properly designed digital filters. Each radiating sound source or hole requires a first- or second-order digital filter to account for angular- and frequency-dependent pressure distribution characteristics. Sound propagation delay from the source to the pickup is modeled with a delay line and possibly a fractional-delay interpolation filter. An additional digital filter to model attenuation in free space can also be used. The results of this model compare well with frequency-domain polar radiation calculations and measurements performed by Antoine Rousseau and René Caussé (1996) at the Institut de Recherche et Coordination Acoustique/Musique (IRCAM). A simplified system appropriate for real-time synthesis was developed using The Synthesis ToolKit (STK) that allows continuous pickup movement within an anechoic 3D space.

Current efforts are directed toward the development of simplified models of conical woodwind instruments and their excitation mechanisms [Scavone, 2002]. This work is being performed in conjunction with the *Categorical Perception of Sound Sources* project, which is described in the *Psychoacoustics and Cognitive Psychology* research section of this document.

References:

- Scavone, G. P. and Karjalainen, M. (2002) *Tonehole Radiation Directivity: A Comparison Of Theory To Measurements*, Submitted to the 2002 International Computer Music Conference.

- Scavone, G. P. (2002) *Time-Domain Synthesis of Conical Bore Instrument Sounds*, Submitted to the 2002 International Computer Music Conference.

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6.2.6 Analysis and Synthesis of Unusual Friction-Driven Musical Instruments

Stefania Serafin, Patty Huang, Solvi Ystad, Julius Smith, and **Chris Chafe**

The focus of this research is to model different musical instruments whose main excitation mechanism is driven by friction. Friction is the excitation source of a well-known family of musical instruments, i.e. the family of bowed string instruments. Friction, however, also represents the excitation source for less common instruments such as the glass harmonica and the musical saw.

In this work, we use the idea that musical instruments can be decomposed into an exciter and a resonator. The exciter in all these case is the driving friction mechanism, while the resonator is either a wineglass or a saw blade.

All these models are implemented in real-time using Pd.

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- Ystad, S. (2000) *Sound Modeling Applied to Flute Sounds*, Journal of the Audio Engineering Society, Vol. 48, No. 9, pp. 810–825, September 2000.

6.2.7 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 partially 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, as well as other nonlinearities. 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:

- J. O. Smith, *Digital Waveguide Modeling of Musical Instruments*, `http://www-ccrma.stanford.edu/~jos/waveguide/`.
- J. O. Smith, "Physical Modeling Synthesis Update", *Computer Music Journal*, vol. 20, no. 2, pp. 44–56, Summer, 1996. Available online at `http://www-ccrma.stanford.edu/~jos/pmupd/`.
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- J. O. Smith, *Introduction to Digital Filters*, `http://www-ccrma.stanford.edu/~jos/filters/`.

6.2.8 Applications of Bioacoustics to the Creation of New Musical Instruments

Tamara Smyth

Animal sound production mechanisms are remarkably similar to those of many musical instruments. Though vibrating membranes, plates and shells, and acoustic tubes and cavities, are important components of any acoustic system, in musical and bioacoustic systems these components enable the production of sound that is undeniably captivating to human listeners.

There are, however, intriguing and potentially musical sounds produced by bioacoustic mechanisms that do not exist in traditional musical instruments. This research concentrates on these particular bioacoustic systems, and through the development of mechanical and computational models, aims to determine whether any aspect of the system is suitable for musical instrument design (either in sound production, acoustic output, or user control).

There are two aspects of instrument design that are being addressed: sound production, which involves the use of physical modeling techniques to develop quality sound synthesis models of animal sound production mechanisms, and sound control, building haptic human interfaces to the computer synthesized instrument, based on these same mechanisms.

A mechanical model of the cicada's unique and efficient sound excitation mechanism was built to determine whether or not such a mechanism could also be used by a human (who has more limited muscular control). In addition to providing an accurate input signal (one that represents the buckling ribs of the cicada) to the physical model, it serves as a mechanical haptic interface, facilitating the user's ability to control the instrument's sound. The mechanical controller is being used for both scientific understanding of the cicada's buckling mechanism and for musical experimentation.

Another bioacoustic system currently being researched is the bird's syrinx. This system is of particular interest because, in addition to being a musical inspiration to many composers, it has a unique structure that allows for rapid shifts from low to high registers in the bird's, often virtuosic, song.

Far too often, parameter-rich physical models are developed with no means of controlling them. Likewise, musical controllers are often built with nothing to control. In both cases, the music is lost. This research intends to bridge the separation between the development of parametric sounds and the development of the devices used for controlling them, while offering new and intriguing musical instruments to contemporary musicians.

Reference:

- Smyth, T. and Smith, J. O. (2002) *Creating Sustained Tones with the Cicada's Rapid Sequential Buckling Mechanism.* Proceedings of the 2002 Conference on New Instruments for Musical Expression (NIME-02), Dublin, Ireland, May 2002.

6.3 Digital Signal Processing

6.3.1 Efficient Pitch Detection Techniques for Interactive Music

Patricio de la Cuadra, **Aaron Steven Master**, and **Craig Stuart Sapp**

Several pitch detection algorithms have been examined for use in interactive computer-music performance. We define criteria necessary for successful pitch tracking in real-time and survey four tracking techniques: Harmonic Product Spectrum (HPS), Cepstrum-Biased HPS (CBHPS), Maximum Likelihood (ML), and the Weighted Autocorrelation Function (WACF).

Reference:

- Cuadra, P., Master, A., and Sapp, C., *Efficient Pitch Detection Techniques for Interactive Music*, Proceedings of the 2001 International Computer Music Conference, Havana, Cuba, pp. 403–406. Computer Music Association. Available online at `http://www-ccrma.stanford.edu/~craig/papers/01/icmc01-pitch.pdf`.

6.3.2 Multichannel Acoustic Echo Cancellation for Telepresence Applications

Yi-Wen Liu

With the availability of increased communication bandwidths in recent years, people have become interested in full-duplexed, multichannel sounds for telepresence services because of the potentials for providing much better hearing experiences. Nevertheless, in any full-duplex connection of audio network, the problem of acoustic echoes arises due to the coupling between loudspeaker(s) and microphone(s) placed in the same room. Moreover, it is known that the cancellation of multichannel acoustic echoes is a mathematically ill-conditioned problem. What happens is that, due to the high correlation between signals in multiple channels, an adaptive echo canceller tends to converge to a degenerate solution and fails to find the true coupling paths.

Although several types of algorithms for decorrelating the channels have been proposed to regularize the problem in the context of speech teleconferencing, these algorithms are all developed based on the criterion that any sound other than the speech signals should not be heard. However, it is not necessarily so in applications such as video games and performing arts where background sound effects and background music are common and desired practices.

I am currently working on utilizing background sounds for multichannel echo canceling. Methods are developed to generate arbitrarily many orthogonal and perceptually similar sounds from a mono source, and the sounds are fed into a multichannel echo canceler for the canceler to better identify the echo paths.

Through the AEC experiments that have been conducted, it is found that there are tunable parameters in both the sound pre-processing stage and the adaptive learning stage. While the parameters are selected in an *ad hoc* way presently, in the future, I would like to pursue the direction of formulating the selection of parameters as an optimization problem. Also, another interesting domain is to design an adaptive learning alrogithm that is customized to the AEC problem, possibly with some physical knowledge incorporated.

Reference:

- Liu, Y. and Smith, J.O. (2002) *Perceptually similar orthogonal sounds and applications to multichannel acoustic echo canceling*, Proceedings of AES 22nd International Conference, Espoo, Finland (June 2002). Available online at `http://www.stanford.edu/~jacobliu/papers/`.

6.3.3 Nonstationary Sinusoidal Modeling

Aaron Master

We present a new analysis methodology for extracting accurate sinusoidal parameters from audio signals. The method combines modified vocoder parameter estimation with currently used peak detection algorithms in sinusoidal modeling. The current system processes input frame by frame, searching for peaks like a sinusoidal analysis model, but also dynamically selects vocoder channels, through which smeared peaks in the FFT domain are processed. This way, frequency trajectories of sinusoids of changing frequency within a frame may be accurately parametrized. We note that the computational expense incurred by using the new model is offset by the reduced frame rate allowed, and describe possible applications for the new model. We demonstrate that the current model is able to follow the changing frequencies in a long analysis frame, with accuracy greater than that found in a conventional sinusoidal model.

Reference:

- Aaron Master, *Nonstationary Sinusoidal Modeling.* Proceedings of the 2002 IEEE International Conference on Acoustics, Speech, and Signal Processing, Orlando, FL.

6.3.4 Scanned Synthesis

Max V. Mathews, **Bill Verplank**, and **Rob Shaw**

"Scanned synthesis" is done by scanning the slowly varying shape of an object and converting this shape to samples of a sound wave. The shape of the object is determined by the dynamic reactions of the object to forces applied by the performer. These forces vary at "haptic" rates (0–20 Hz). If the scanning path is closed, the sound wave is quasiperiodic and a fundamental pitch is perceived at the cycling frequency (20 Hz–20 kHz). Scanned synthesis provides direct dynamic control by the performer over the timbre of sounds as they are produced. The object can be real or simulated. With finite-element models, we have simulated the one-dimensional wave equation for a generalized slowly vibrating string. Timbres generated by manipulating the string at haptic rates are perceived as having a very pleasing live quality caused by the continually changing spectrum. To achieve additional richness, the performer can change the properties of the string in time and over the length of the string.

6.3.5 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 in the past decade or so 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 useful 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 requires 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.*

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 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, (3) noise reduction, and (4) novel synthesis systems based on the direct creation and transformation of spectral events and envelopes.

References:

- Levine, S., *Audio Representations for Data Compression and Compressed Domain Processing*, Ph.D. thesis, Electrical Engineering Department, Stanford University (CCRMA), December 1998. Available online at `http://www-ccrma.stanford.edu/~scottl/thesis.html`.
- Serra, X., and J. O. Smith, "Spectral Modeling Synthesis: A Sound Analysis/Synthesis System Based on a Deterministic plus Stochastic Decomposition," *Computer Music J.,* vol. 14, no. 4, pp. 12–24, Win., 1990. The latest free Spectral Modeling Synthesis (SMS) software can be obtained from the SMS home page at `http://www.iua.upf.es/~sms`.
- Smith, J. O., "Music 420 (EE 367A) Bibliography: Audio Spectral Modeling," `http://www-ccrma.stanford.edu/CCRMA/Courses/420/References.html`.
- Smith, J. O., *Mathematics of the Discrete Fourier Transform*, `http://www-ccrma.stanford.edu/~jos/mdft/`.

6.3.6 Transient Detection and Modeling

Harvey Thornburg

Transient events are regions not "well-modeled" by a locally stationary sinusoidal model. Examples include abrupt changes or fast decays/modulations in mode amplitudes/frequencies. Transient regions usually follow onsets, which are preceded by an abrupt change. This suggests a twofold approach for transient detection:

- *Segmentation:* Find boundaries of abrupt change in the underlying signal model (e.g. sinusoidal, Gaussian AR, or some physically-informed model)
- *Transient Characterization:* Classify each region depending on some predetermined cost criteria (e.g. expected bitrate).

Segmentation – Bayesian Approach:

- The classical approach concerns a piecewise constant signal model, with statistically independent segments. We have recently adopted a more general, unified Bayesian framework, as follows: Let $\delta_t \in \{0,1\}, \delta_t = 1$ indicate change at time t. Then, the model for the signal y_t is described:

$$\begin{aligned}\theta_{t+1} &= \begin{cases} \theta_t + u_t & \delta_t = 0 \\ v_t \sim p(v_t|\theta_t) & \delta_t = 1 \end{cases} \\ y_t &\sim f(y_t|\theta_t, y_{1:t-1}), t \in \{1 \ldots T\}\end{aligned}$$

- This framework allows us to exploit additional prior information and information about musical structure, according to the specifications:
  - *Prior probability of change*: The distribution $p(\delta_{1:T})$ may encode information about the structure of rhythm.
  - *Markov evolution of parameter jumps*: The distribution $\{p(\theta_0); p(v_{1:T}|\theta_{1:T})\}$ may encode information about melodic/timbral evolution.
  - *Allowance for slow parameter variations*: The distribution $p(u_t)$ may be used to allow for slow, continus variations in the model parameter within a segment, to emphasize the "abruptness" of change.

Applications:

- *Joint Rhythm Tracking and Onset Detection:*
  - A three-layer switching state space model (rhythm tracker) is used to learn the pattern of onsets. The top layer S_t encodes the discrete rhythmic interval and metrical position; middle layer X_t encodes tempo and inherent onset position; bottom layer y_t gives the observed onset position.
  - The rhythm tracker produces the posterior distributions about the next segment points given segments observed so far, e.g. $P(X_{t+k}|y_{1:t}), k \in \{1, 2, \ldots\}$ The segmenter uses these distributions as local priors to detect the next batch of segments, which provide subsequent observations for the rhythm tracker. The net effect is improved segmentation performance about musically relevant changes (onsets); spurious changes are ignored or suppressed. The behavior mimics somewhat the cognitive activity of the human listener, though rigorous parallels are not yet established.
- *Harmonic Comb Models for Piano Transcription:* To improve segmentation performance for specific musical signals, we wish to exploit a higher degree of structure than is available from generic (unconstrained) AR or sinusoidal models. Additional structure allows us to support a high model order, or high number of sinusoids modeled, because the model is highly constrained in a probabilistic sense. The additional structure may be motivated by explicit knowledge of the physics of a particular instrument, say, piano.
- *Changeograms:* The changeogram gives a nonparametric view of the posterior probability that an abrupt change occurs at a particular time based only on information in local windows. Uses and properties are as follows:
  - The changeogram may be peak-picked/thresholded to yield a "quick and dirty" estimation of change points
  - The changeogram itself serves as an "empirical Bayes" prior for further offline Bayesian segmentation. The inherent structural assumption is that changes are infrequent but occur in clumps.

- Changes spaced far enough apart with respect to the window size appear resolved as "peaks" in the representation. The height of the peaks corresponds to the intensity of the change.
- The size of the window limits resolution: When two changes are spaced at less than the window size, the change with less intensity does not survive.
- A kernel may be chosen such that peaks are dilated in the representation. For the "empirical Bayes" approach, the kernel expresses uncertainty that additional change points have been masked by the main peaks.

6.4 Controllers and Musical Instruments

6.4.1 TouchSound: Haptics in Sound Editing

Lonny Chu

Recent studies in haptics have shown that force-feedback interfaces can improve user efficiency and accuracy while decreasing the cognitive load required to accomplish computer tasks. These types of results can be greatly beneficial to music as we strive to create interfaces that allow the user to become immersed in the musical experience without being overly conscious of specific physical gestures. While current sound editing systems require the musician to use devices such as keyboards and mice, passive scroll wheels, or passive joysticks while editing sound, TouchSound uses a force-feedback mouse, a vibrotactile mouse, and a force-feedback knob to investigate how programmable forces can improve the sound editing experience.

The two primary goals of TouchSound are:

1. Show that force-feedback interfaces improve user performance in editing sound.
2. Explore the design processes necessary for creating pertinent haptic effects, or haptic icons, that will assist the musician in using the environment.

For the first goal, experiments will be performed to measure performance in basic sound editing tasks such as locating the onset and offset of a sound sample. Various haptic effects such as detents, pops, textures, walls, and damping will be used to construct the haptic environment as the user is tracked in accomplishing tasks such as locating defined points in the sound. Additionally, subjective data will be collected to show that haptics also increases user fulfillment and decreases stress levels. Future work will then investigate issues involving the design of haptic icons for artistic purposes.

6.4.2 The Accordiatron: A New Gestural MIDI Controller

Michael Gurevich

The Accordiatron is a MIDI controller for interactive performance based on the paradigm of a conventional squeeze box or concertina. It senses and encodes the gestures of a performer using the standard communication protocol of MIDI, allowing for flexible mappings of performance data to sonic parameters. When used in conjunction with a real-time signal processing environment, the Accordiatron can become an expressive, versatile musical instrument. It features a combination of discrete and continuous sensory data, providing the subtle expressiveness and control necessary for interactive music.

6.4.3 The vBow: A Four Degrees of Freedom Haptic Virtual Violin Bow Controller

Charles Nichols

The vBow, a virtual violin musical controller human computer interface, has been designed to provide the computer musician with most of the gestural freedom of a bow on a violin string. Four cable and servomotor systems allow for four degrees of freedom, including the lateral motion of a bow stroke across a string, the rotational motion of a bow crossing strings, the vertical motion of a bow approaching and pushing into a string, and the longitudinal motion of a bow traveling along the length of the string. Encoders, attached to the shaft of each servomotor, sense the gesture of the performer, through the rotation of the servomotors driven by the motion of the cables. The data from each encoder is mapped to parameters in synthesis software of a bowed string physical model. The software also sends control data to the servomotors, engaging them and the cables attached to them with a haptic feedback simulation of vibration, friction, and elasticity.

For more information, visit `www.CharlesNichols.com/vBow.html`.

6.4.4 THE PLANK: simple force-feedback

Bill Verplank, Michael Gurevich, and Max Mathews

Active force-feedback holds the potential for precise and rapid controls. A high performance device can be built from a surplus disk drive and controlled from an inexpensive microcontroller. Our new design, *The Plank*, has only one axis of force-feedback with limited range of motion. It is being used to explore methods of feeling and directly manipulating sound waves and spectra suitable for live performance of computer music, in particular, scanned synthesis.

Reference:

- Bill Verplank, Michael Gurevich, Max Mathews, *THE PLANK: Designing a simple haptic controller.* Proceedings of the Conference on New Instruments for Musical Expression (NIME-02), Dublin, Ireland, May 24-26, 2002.

6.4.5 Designing Controllers: The evolution of our Computer-Human Interaction Technology Course

Bill Verplank

Over the last several years, with support of the CS department, we have developed a series of lectures, labs and project assignments aimed at introducing enough technology so that students from a mix of disciplines can design and build innovative interface devices. We have come to focus less on theory and more on practical skills leading to a four-week project: designing and building a working controller.

6.4.6 The Mutha Rubboard Controller

Carr Wilkerson, Carmen Ng, and Stefania Serafin

The Mutha Rubboard is a musical controller based on the rubboard, washboard or frottoir metaphor commonly used in the Zydeco music genre of South Louisiana. It is not only a metamorphosis of a traditional instrument, but a modern bridge of exploration into a rich musical heritage. It uses capacitive and piezo sensing technology to output MIDI and raw audio data.

This new instrument is meant to be easily played by both experienced players and those new to the rubboard. It lends itself to an expressive freedom by placing the control surface on the chest and allowing the hands to move uninhibited about it or by playing it in the usual way, preserving its musical heritage.

Visit `http://www-ccrma.stanford.edu/~carrlane/mutha/muthahome.html` for more information.

Reference:

- Wilkerson, C., Ng, C., and Serafin, S. (2002) *The Mutha Rubboard Controller.* Proceedings of the 2002 Conference on New Instruments for Musical Expression (NIME-02), Dublin, Ireland, May 2002.

6.5 Audification of Data

6.5.1 Auditory Representation of Complex Data

Jonathan Berger, **Michelle Daniels** and **Oded Ben Tal**

We describe our current research on the auditory representation of complex data in which we sonify multidimensional data using an filterbank with noise or pulse train input with the goal of creating an intuitive, easy to learn representation of multiple, simultaneous independently changing parameters. Preliminary experiments suggest a promising model using a subtractive synthesis approach. Our sound examples include sonification of data acquired by marine scientists measuring the salinity and temperature at various depths in the Dead Sea.

The vowel like sounds produced by the filter instrument provides an intuitive point of reference which can be used to measure changing states of data. We noted that fluctuations of dynamic envelope control of center frequency and bandwidth in multiple simultaneous data sets each set with individual components in discrete frequency ranges provide a recognizable auditory representation of the overall trends of each individual data set. The research is supported by the Stanford Humanities Laboratory.

References:

- Ben-Tal, O., Berger, J. and Daniels, M. (2001) *De Natura Sonoris: Sonification of Complex Data,* Proceedings of the WSEAS Conference on Mathematics and Computers in Science and Engineering.
- Ben-Tal, O., Berger, J. and Daniels, M. (2001) *Sonification in Science and Art,* Proceedings of the 2001 ESCOM Conference, Liege, Belgium.

6.5.2 SonART - The Sonification Application Research Toolbox

Oded Ben Tal, **Jonathan Berger**, **Bryan Cook**, **Michelle Daniels**, **Gary Scavone**, and **Perry Cook**

The Sonification Application and Research Toolbox (SonART) is an open-source effort whose core code is platform-independent. The primary objective of SonART is to provide a set of methods to map data to sonification parameters along with a set of graphical user interface tools that will provide practical and intuitive utilities for experimentation and auditory display. SonART provides publicly available, well-documented code that is easily adapted to address a broad range of sonification needs. The effort builds upon the Synthesis ToolKit in C++ (STK) (Cook and Scavone, 1999), both of whose authors are part of this research effort.

By classifying sonification methods, the SonART provides researchers with the means of exploring parameter mapping with the same high level control afforded by many data visualization packages. Synthesis and sound processing parameters can be classified by general acoustic or musical properties or by synthesis specific parameters.

Parameter mapping using general acoustic properties or synthesis specific parameters is potentially limited by two factors. First, the finite number of parameters used in a given synthesis method limits the dimensionality of the data to be sonified. Second, the mapping of data to a particular parameter may not be intuitive to the data analyst. One approach to address the difficulty of intuiting the meaning of sonified data involves using sounds resembling those in nature. Synthesis techniques that approximate natural sounds, and physical models that simulate material interactions (such as springs and dampers), may prove useful parameter mapping techniques. Instead of mapping a data dimension to an arbitrary synthesis parameter, a mapping that produces intuitive natural sounds such as vowels (Ben Tal, Berger and Daniels, 2001) may be used. Using this approach, vowel quality or the proximity of a sound to a cardinal vowel can provide an intuitive basis for sonification. While natural sounds may provide more intuitive and more easily interpreted results, they introduce complications in terms of how parameter mapping methods are organized. Physical models can be used in such a way that the data miner interacts with data by excitation of a sound that impinges upon data points.

6.5.3 SoundWIRE: Sound Waves on the Internet from Real-time Echoes

Chris Chafe, Scott Wilson, Randal J. Leistikow, Gary P. Scavone, Daniel Walling, Nathan Schuett, Christopher Jones, and **David Chisholm**

New, no compromise, computer applications for audio have been demonstrated using a simplified approach for high quality music and sound streaming over IP networks. Audio is an unforgiving test of networking - if one data packet arrives too late and we hear it. Traditionally, compromises of signal quality and interactivity have been necessary to avoid this basic fact. Along with our new professional audio applications we have developed SoundWIRE, a utility which affords an intuitive way of evaluating transaction delay and delay constancy. Its final form is an enhanced "ping" that uses actual sound reflection. A musical tone, such as a guitar pluck, can be created by repeatedly reflecting a digital acoustic signal between two hosts. Using the network delay between these reflections to substitute for a guitar string creates a tone whose stability represents perfectly regular service and whose pitch represents transmission latency. The ear's ability to discern minute differences makes this an unforgiving test of network reliability.

References:

- Chafe, C. and Leistikow, R. (2001) *Levels of Temporal Resolution in Sonification of Network Performance.* Proceedings of the 2001 International Conference On Auditory Display, Helsinki, Finland.

- Chafe, C., Wilson, S., Leistikow, R., Chisholm, D., and Scavone, G. (2000) *A simplified approach to high quality music and sound over IP.* Proceedings of the 2000 International Conference On Digital Audio Effects, Verona, Italy.

6.6 Techniques and Approaches for Computer-Music Composition

6.6.1 Signal Processing Techniques for Algorithmic Composition

Christopher Burns

This project is oriented towards the development of new tools for algorithmic composition, based on traditional signal processing techniques. While we ordinarily associate filtering and frequency transforms with sampled audio, these methods also possess a number of desirable properties for the generation of higher-level musical materials. These techniques offer meaningful and intuitive relationships between input and output. Additionally, many such tools have "strong parameters," where a change to a single parameter produces a substantial and observable alteration to the output. Finally, the notion of "frequency," abstracted to rate of change, analogizes well to music. Harmonic rhythm is the most obvious example, but music in general is multitemporal; operating on a number of different time scales simultaneously, from notes and phrases to sections, movements and complete works.

The first application of these tools was in a work for violin solo titled *Integrities*. In this instance, the time-domain outputs of filters were mapped to musical parameters. Over the course of the piece a number of different filters were used, with particular emphasis on time-variant resonators displaying "classic" behaviors like sweeping filter resonance or bandpass frequency. A variety of different mappings were also tested over the course of the piece, including inter-event onset times, phrase onset times, phrase durations, and pitch.

Current work on a companion piece for cello solo concentrates on the sonification of frequency transforms. Spectrograms of speech recordings and other structured audio are the principal data source, and mappings include pitch selection, event duration, and dynamics.

6.6.2 Parameter Manipulation for Composing with Physical Models

Juan Reyes

The problem with physical models and their appeal to composers is not merely perceptual or aesthetic. Furthermore, it is not a question of understanding the physics and parameters of the actual instrument. It is a question of achieving musical textures, realistic articulations and extending the qualities of a family of sounds given by a single characteristic timbre in a physical model of an instrument. This can be achieved by modeling and manipulating expression parameters for musical gesture. We are dealing with compositional techniques that render a computer music piece using physical models as the primary source technique for sound synthesis in non-realtime environments and parameter manipulation by means of envelopes, randomness and chaotic signals for expressiveness.

6.6.3 Composing with the Physical Model of the Maraca

Juan Reyes

The physical model of the maraca is a very flexible algorithm for generating interesting timbres out of the percussion family of instruments. It is also well suitable for achieving musical expression in the digital synthesis of a sound. A piece called *Wadi Musa* (or *The Monteria Hat*) was composed using direct digital synthesis from the physical model of the maraca in the Common Lisp Music (clm) environment. The original physical model was developed by Perry Cook as part of his PhISM approach for using computer models of percussion instruments. The clm version is a direct transcription of the Synthesis ToolKit (STK) algorithm done by Bill Schottstaedt. A variety of parameters and algorithms were used to achieve interesting musical structures in a composition for which the "maracas" sound is the underlying musical element in a piece for tape and live instruments. Performance modeling follows research started also by Perry Cook, Brad Garton, Chris Chafe and others around 1995. Although physical models have not proven to be appealing to composers because of their nature of imitating real world phenomena, in this case the model of the maraca was proven to be a good tool for achieving a variety of creative goals in pursuing a new music composition. Basic parameters such as note duration and dynamic range produce a wide range of musical material and perhaps in new aesthetic in the perception of computer generated music.

6.7 Psychoacoustics and Cognitive Psychology

6.7.1 Categorical Perception of Sound Sources

Stephen Lakatos, Gary P. Scavone, and James Beauchamp

The human auditory system possesses a remarkable ability to differentiate acoustic signals according to the vibrational characteristics of their underlying sound sources. Understanding how listeners can detect, discriminate, classify, and remember acoustic source properties forms this project's long-range goal.

The present project brings to bear on these topics techniques of psychophysical measurement, spectral analysis/synthesis techniques, and computer simulation of acoustic objects. Using such interdisciplinary approaches, studies will determine the validity of a three-stage model of auditory source perception:

1. an initial stage that segregates sounds according to basic spectral and temporal features
2. a second stage that parses the vibrational modes of their underlying sound sources
3. a third stage that integrates the vibrational modes across various acoustic contexts and generates a source representation that is invariant across a broad range of sounds

Using methods of signal detection, preliminary studies will determine how listeners' sensitivity to auditory signals depends on whether attention is first directed to their acoustic features, and how sensitivity may improve as a function of the available source cues. Additional studies will use physical modeling and spectral simplification techniques to determine which acoustic features are critical to detection performance. A fundamental problem in auditory perception is to understand how listeners can perceive a sound source to be constant across wide variations in the range of sounds that the source can produce. Consequently, a separate set of studies will use adaptation techniques to determine how listeners categorize sounds by their source characteristics, and to assess whether computer-generated prototypical sources – sources, such as bars, tubes, and plates, that define broad classes of sound-producing objects – are classified more rapidly and accurately than non-prototypical sources. Our ability to recognize previously heard sounds suggests that we encode features of acoustic sources in memory. A related set of experiments will use recognition and recall tasks to determine what features of sounds are encoded in working and long-term memory, and whether memory representations encode a sound's surface spectral-temporal features or its underlying physical source characteristics.

In sum, this research program should shed important light on the representation of auditory source characteristics by determining the stages of processing that auditory information undergoes from its initial encoding at peripheral levels to its source-based representation at more central levels. Not only can this improve our basic understanding of auditory processing but also can suggest ways in which humans can optimize their performance in detecting and evaluating signals of interest within their acoustic environment.

References:

- Scavone, G. P., Lakatos, S, Cook, P. R., and Harbke, C. (2001) *Perceptual spaces for sound effects obtained with an interactive similarity rating program.* International Symposium on Musical Acoustics, Perugia, Italy. September 2001.
- Lakatos, S., Cook, P. R., and Scavone, G. P. (2000) *Selective attention to the parameters of a physically informed sonic model,* Acoustic Research Letters Online, Acoustical Society of America, March 2000.

6.7.2 Potential Applications of Linguistic Theory to Music Theory

Leigh VanHandel

This research explores the possible application of recent developments in the transcription and study of "intonation" in linguistics to music theory and analysis. In the field of linguistics, "intonation" refers to the "melody" of an utterance, including such characteristics as pitch, stress, accent, and phrasing.

In recent research by Pierrehumbert (1980) and Beckman and Elam (1997), among others, an intonation transcription method known as ToBI (Tone and Break Indicies) has been developed and codified. This system has become essentially a standard transcription technique for English dialects.

Using the basic foundation of ToBI transcription, I researched possible applications of the theory to musical analysis and perception. Strengths and weaknesses of the application of the theory to music were explored, as well as potential limitations to applicability, including stylistic elements and genre.

6.8 Machine Recognition in Music

6.8.1 Koto Musical Score Database

Sachiko Deguchi and **Craig Stuart Sapp**

Research is being done on the representation of koto musical scores in electronic format for use in music analysis with computers. The koto is a Japanese instrument with 13 plucked strings. Each string can be tuned to any pitch by positioning the bridge of each string, although tunings used in traditional koto music are limited. Koto music was originally transmitted orally, using syllables to indicate basic musical patterns, but not exact pitches. Western music notation is not sufficient to describe the numerous ornamental techniques, so modern koto players have developed their own notation systems based on numbers for each string plucked by the right hand and a set of modifying symbols for the pitch ornamentations done primarily played with the left hand. Converting these scores to Western notation is not sufficient for analysis by computer because much of the ornamental techniques are unique to koto playing. Sample musical encodings of koto scores are available on the web at `http://www-ccrma.stanford.edu/~craig/koto`.

6.8.2 Automatic Transcription of Polyphonic Piano Music

Randal J. Leistikow

The goal of this project is to develop a system that accepts as input a single-channel or stereo recording of solo piano music and outputs a data file containing the performance parameters necessary to resynthesize an expressively realistic performance of the music. Applications of such a system include the abilities to "resurrect" performances from historic recordings and hear them played live on a modern reproducing piano, release pristine versions of recordings corrupted by noise, process the output data to study aspects of performance practice or playing style, and code piano music extremely efficiently.

6.8.3 Computational Models for Musical Style Identification

Yi-Wen Liu and **Craig Sapp**

Research is underway to identify musical features which can be used to distinguish between different composers writing in a common style. In the preliminary experiments on Mozart and Haydn's string quartet minuets, probabilities of transition between classes of musical events (such as pitch classes, rythmic classes, etc.) are computed and compared in the information theoretic sense. For instance, an experiment showed that the Kullback-Leibler distance is 57% "successful" (8/18 in Mozart and 34/50 in Haydn) on the Haydn versus Mozart minuet identification based on the note transition probabilities of the first violin part.

As a control for the identification accuracy of computational models, a human-based experiment is being conducted over the web at `http://qq.themefinder.org` where randomly selected MIDI files of string quartet movements composed by either Mozart or Haydn are played to listeners. The test takers must then choose the composer who they think wrote the musical sample being played. Summary statistics for identifications are viewable in real-time on the experiment's webpage. For example, a prototypical Mozart composition would be K 456, movement 4, where 100% (6/6) correct identifications have been made. A prototypical Haydn quartet movement would be Op. 50, no. 3, movement 1, which so far has 100% (5/5) correct identifications. For all test takers, the average accuracy of distinguishing between Mozart and Haydn string quartets is 60% over 470 trials. Trained classical musicians with no experience listening to the string quartets of either Mozart or Haydn can identify the correct composers with about 70% accuracy.

Musical data for the experiments has been provided by CCARH which has electronically encoded the scores for nearly all of Mozart and Haydn's string quartets.

6.8.4 Harmonic Visualizations of Tonal Music

Craig Stuart Sapp

Multi-timescale visualization techniques for displaying the output from key-finding algorithms have been developed for harmony analysis in music. The horizontal axis of the key graphs represents time in the score, while the vertical axis represents the duration of an analysis window used to select music for the key-finding algorithm. Each analysis window result is shaded according to the output key's tonic pitch. The resulting diagrams can be used to compare differences between key-finding algorithms at different time scales and to view the harmonic structure and relationships between key regions in a musical composition. Example plots are available on the web at: `http://www-ccrma.stanford.edu/~craig/keyscape`.

Reference:

- Craig Sapp, *Harmonic Visualizations of Tonal Music.* Proceedings of the 2001 International Computer Music Conference, Havana, Cuba, pp. 423–430. Computer Music Association. Available online at http://www-ccrma.stanford.edu/~craig/papers/01/icmc01-tonal.pdf.

6.8.5 Themefinder: A Musical Theme Search Engine

Craig Stuart Sapp

Suppose you have a melody stuck in your head, but you don't know the name of it. You can now search a collection of over 36,000 musical themes on the web at `http://www.themefinder.org` in an attempt to identify it. Musical themes can be searched with different levels of exactness, going from a precise sequence of pitch names to basic melodic contours. Wildcards similar to those used in regular expressions are supported in most types of searches. Themefinder is useful for research purposes as well. It has been used to identify melodies suitable for musical performance experiments and has also been used to idenitfy common starting pitch patterns in music. *Themefinder* is a collaboration between the Center for Computer Assisted Research in the Humanities at Stanford University and the Cognitive and Systematic Musicology Laboratory at Ohio State University. *Themefinder* uses the Humdrum data format for encoding and manipulating music for search and display on the website.

6.8.6 Audio Content-Based Retrieval Methods and Automatic Style Classification

Unjung Nam

The rapid proliferation of user accessable digital music data poses significant challenges to the tasks of searching and retrieving music from massive data sets. Signal analysis methods to derive high level musical information such as pitch and instrumentation could be used to formulate classification methods that may prove useful in efficient and flexible content retrieval. My research evaluates some current approaches to content-based music retrieval methods and proposes a model that attempts to distinguish among three classes of music (jazz, popular, and classical) by analysis and extraction of features in the frequency and time domains.

6.9 Historical Aspects of Computer Music

6.9.1 New Realizations of Electroacoustic Works

Christopher Burns

There are a number of reasons to create new realizations of favorite electroacoustic works. First and foremost are the reasons for performing any interesting piece of music. Performance creates the opportunity to share the work with new audiences, and encourages close study of the work by the performer. This engagement is especially important for indeterminate or otherwise flexible works which require the performer to make decisions traditionally considered compositional. Additionally, many electroacoustic works will eventually require rescue from technological obsolescence. New realizations, using new technologies, can extend the performing lifespan of a piece with complex technical requirements. Finally, the process of realization admits the possibility of an evolving, performing tradition for a particular work, with new solutions and interpretations enriching the music s sense of possibility.

Two recent realizations by the author are useful case studies in the creation of new performing versions of electroacoustic music. Although very different works, Alvin Lucier's *I am sitting in a room* and Karlheinz Stockhausen's *Mikrophonie I* present some similar challenges in realization. Both works have relatively open, flexible scores which encourage experimentation and variation. They also have well-established performing traditions, centered on the composer as authoritative interpreter, which have minimized the flexibility suggested by the the variations and alternatives possible in Lucier and Stockhausen s music.

Our experiences realizing and performing *I am sitting in a room* and *Mikrophonie I* suggests that the interpretive aspects of a realization are not established in a single moment, but are the product of a series of small decisions and practical solutions - as is the case with most musical performances. Every question must be met with an appropriate balance of textual fidelity, musical effectiveness, and pragmatism.

Continuing work on this project focusses on the creation and performance of additional realizations: recently completed works include *Rozart Mix* (1965) by John Cage, *Poème Symphonique* (1962) by György Ligeti, and *Study #21 (Canon X)* for player piano (ca. 1950s) by Conlon Nancarrow.

6.9.2 Compositional Process and Documentation in Computer Music

Christopher Burns

As more and more composers produce electronic records, musicians and scholars will have to come to grips with these new forms of documentation. In some ways the new forms will be preferable: electronic records can be very revealing about a composer's technical intentions. What a composer can scribble on paper in a cryptic shorthand has to be spelled out in detail to make software work; composers' code and other electronic records may prove more useful than paper sketches when working methods are complex. However, these sketches are only comprehensible if the reader is initiated into the programming language, and only fastidious and self-conscious composers are likely to preserve their mistakes, false starts, and reconsiderations, through multiple versions of their code.

This project considered the sketches and other creative intermediates from Fernando Lopez-Lezcano's *iICEsCcRrEeAaMm* for four-channel tape, and Christopher Jones' *Matragn* for clarinet and CD. The sketches for *iICEsCcRrEeAaMm* were entirely in electronic form, and consisted principally of software "instruments" and "score" written in the Common Lisp Music language. In particular, repeated revisions of the composer's "grani" granular synthesis instrument and the "dlocsig" spatialization instrument are suggestive of the ways in which compositional desires influenced the software development.

In *Matragn*, the creative work was more focused on pencil sketches, with the programming work coming last. Most of the precompositional work documented in the sketches is borne out in the finished piece. There are a number of revisions and rethinkings of the pitch scheme in the sketches; this seems to have been an area of concern to the composer, and an area for experimentation. The draft of the clarinet part included impressionistic hieroglyphs suggesting possibilities for the sounds on CD. However, comparisons of these graphic sketches with the finished CLM code provide evidence of revisions and rethinkings in the electronic part.

Computer music sketch materials enable us to learn about the composers' creative process, and may in some instances be useful in an analysis of the completed works. It is important to remember, however, that there are a wide variety of working methods, technologies, and records, and some of them are much less genial to examination after the fact. Finally, documentation plays an important role in the creation of musical canons: one of the reasons that Karlheinz Stockhausen's work *Kontakte* is discussed, performed, and taught is that the composer published his exhaustive studio diaries made while creating the tape part. Sketches, electronic or otherwise, play a role in the lifespan of the music itself.

6.10 Computer Assisted Music and Acoustics Research

6.10.1 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.

Its address is:

Braun #129
Stanford University
Stanford, CA 94305-3076
tel. (650) 725-9240
fax (650) 725-9290
Web: `http://www.ccarh.org/`

CCARH staffs a two-quarter graduate course sequence, cross-listed with Computer Science:

Music 253	Introduction to Musical Information	`http://www.ccarh.org/courses/253/`
Music 254	Seminar in Music Representation	`http://www.ccarh.org/courses/254/`

This course sequence is team-taught by Eleanor Selfridge-Field and Walter B. Hewlett. The current teaching assistant is Craig Sapp.

CCARH maintains a lab for applications in music notation, analysis, and sound sequencing. It also maintains some CD-ROM titles related to music research. See `http://www.ccarh.org/software/` and `http://www.ccarh.org/lab/` for more information.

CCARH supports a variety of research projects in data development, access, and query, such as *Musedata* and *Themefinder.*

Publications and Performing Materials:

- *Beyond MIDI: The Handbook of Musical Codes*, ed. Eleanor Selfridge-Field (MIT Press, 1997).
  - Comprehensive coverage of codes used in sound, notation, analysis,and interchange of musical data.
  - Updates at `http://www.ccarh.org/publications/books/beyondmidi/`
- *Computing in Musicology*, ed. Walter B. Hewlett and Eleanor Selfridge-Field. Vols. 1-10 (CCARH): `http://www.ccarh.org/publications/books/cm/`
- *Melodic Similarity* (= CM 11) (MIT Press, 1998): `http://www.ccarh.org/publications/books/cm/vol/11/`
- CM 12: covering XML, NIFF, virtual editions, image reconstruction et al. (in progress)
- Operas and oratorios by G. F. Handel: contact `ccarh@ccrma.stanford.edu`

- Parts: Vivaldi Op. 8 (including "The Four Seasons"): `http://www.ccarh.org/publications/scores/vivaldi/op8/`
- Scores and parts: Bach cantatas (in progress)
- Various occasional publications: `http://www.ccarh.org/publications/`

6.10.2 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 collection, 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, the John Backus Archive, and the John W. Coltman Archive.

Background and History:

The Musical Acoustics Research Library (MARL) has its origin in the vast acoustics research collection of the Catgut Acoustical Society (CAS). These files were assembled over many years by CAS and housed in the home of its founder Carleen M. Hutchins. In the late 1980s, CAS began an effort to establish an appropriate long-term residence for the collection, such that it could serve as a valuable reference source for the musical acoustics community at large. In 1992, the Center for Computer Research in Music and Acoustics (CCRMA) at Stanford University was selected as the repository for this library.

In conjunction with the establishment of the Catgut Acoustical Society Library, CCRMA and CAS encouraged the idea of having the personal archives of Arthur Benade and John Backus at the same site. 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 autumn 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. The most recent addition to MARL, the John W. Coltman Archive, was founded in April 1998. The archives of three of the most prominent wind instrument acousticians of our time, together with the extensive string instrument resources of the Catgut Acoustical Society Library, position MARL as a primary musical acoustics reference source in the world.

Organizational activities of MARL are directed by Gary P. Scavone, with faculty representation by Max V. Mathews and Chris Chafe, the director of CCRMA. MARL is a collective and each member/group representative is encouraged to take part in policy decisions. CCRMA, as an equal partner in MARL, is committed to helping establish the library as an important resource of musical acoustics knowledge for the entire global acoustics research community. A World Wide Web (WWW) site has been created for MARL, which will serve as the primary means for disseminating information to the public about the various collections.

Activities:

The primary ongoing activities of MARL are centered on the development of a uniform databasing system to record the sub-collection catalogue information, as well as the creation of World Wide Web (WWW) pages for the dissemination of the library contents to the global musical acoustics community. The MARL WWW pages currently provide Internet access to overviews of the materials available at CCRMA. When requests for particular documents are received, those documents are being scanned and converted to Portable Document Format (PDF) files using Adobe Capture software and subsequently linked to appropriate locations within the MARL WWW pages. The files at CCRMA are also available for on-site perusal by appointment.

MARL activity is coordinated at CCRMA by Gary P. Scavone and organizational decisions are made by agreement among the representatives of each member collection. Activities are ongoing for the addition of new collections to MARL.

6.10.3 Web-Based Infrastructure for Research and Teaching

Julius Smith

The Web is extremely attractive as a basis for supplementary educational infrastructure. Advantages of Web-based tutorials include:

- Links to related documents anywhere on the Web.
- Live demos (Java applets, sound examples).
- Searchable by topic.

On the negative side, excessive use of hypertext links can be distracting, and following links can even derail and disorient the learner. Good design practices can help minimize this problem, e.g.,

- Links always point to a supporting *definition* or *tutorial*. In other words, the student can be advised to click on a link only if the underlined term is not well understood.
- At any time it is easy to *print* the current document in a form optimized for printing. (Alternatively, a book containing the current material can be ordered.)
- A *local search* facility supports navigation-by-topic in a relatively confined scope.

Signal Processing "Knowledge Webs":

The above design goals have been pursued in the development of a *knowledge web* regarding digital signal processing for music applications.[1] Based on our experience to date, online knowledge webs are best suited for (1) online reference and (2) self-paced study. It is apparent that an effective self-paced curriculum could be developed on the basis of online knowledge webs, particularly with the addition of interactive problem sets.

Web Publishing is On the Way:

Full utilization of the Web in online publishing is barely beginning. However, research has already shown that publications available on the Web tend to be cited more often than publications requiring a trip to the library. Much of what technology does for society is making repetitive tasks faster and more convenient; the far greater convenience of online search and retrieval is indisputable. It can be expected that paper-bound publications will be replaced, one way or another, by online versions.

Web Publishing Tools:

There are several impediments for authors attempting to get their writings onto the Web and *well linked* to supporting materials. One problem is the lack of tools for managing interlinked online documents. To address this issue, tools[2] for conveniently managing online documents generated from LaTeX are being developed. Presently, publications may be generated automatically to the Web in HTML, PDF, and compressed PostScript formats. The footer of every HTML page includes a full bibliographic citation and hyperlinks for downloading either of the two hardcopy formats for printing. Thanks to `latex2html`, every page also contains "navigation links" ('next', 'previous', 'up', 'contents', 'index', and the like), which serve to orient the reader. These features are especially useful when a page is reached via a remote hyperlink, such as from a Web search engine.

Databases for Automatic Linking:

When writing an online document, it is useful to draw upon a collection of *links* to related materials, so that the document will be well connected on the Web. Toward such a link collection, the beginnings of a "global index"[3] of online reference materials pertaining to digital signal processing in music and audio

[1] http://www-ccrma.stanford.edu/~jos/pubs.html
[2] http://www-ccrma.stanford.edu/~jos/webpub/
[3] http://www-ccrma.stanford.edu/~jos/JOSGlobalIndex.html

is under development. In addition to its use in automatic link installation, the global index functions on its own as a kind of hypertext *glossary* for web-resident content in signal processing applied to music and audio.

The Open Dictionary:

Most of links in the global index also appear in the Open Dictionary,[4] a website allowing researchers and educators in related fields to organize their respective links into a global hierarchical encyclopedia. The Open Dictionary is also designed to serve as a basis for automatic link installation, among other functions.

Open Dictionary content is currently dominated by CCRMA Web links. As more "linkable" educational content appears on the Web, the percentage of CCRMA content should decrease, as no one institution can be expected to keep up with the entire world. One project of particular note is the well funded MIT OpenCourseWare (MITOCW) project,[5] in which all MIT courseware is to be placed on the Web for free public access. The second largest source of "definition links" in the Open Dictionary is presently Eric Weisstein's Math World[6] and Treasure Troves[7] of math, physics, astronomy, and music. Due to the excellent "linkability" of Weisstein's online encyclopedias, Math-World and Treasure-Trove definitions blend seamlessly with CCRMA definitions. Longer term, we expect to be linking to best-of-category online tutorials all over the world, as they slowly (so far) come into existence.

Concept Home Pages:

In general, the best link targets tend to be "Concept Home Pages" (CHP), i.e., Web destinations devoted to a *single topic.* Like any good "hub" on the Web, a CHP attempts to efficiently route all types of visitors to all types of content on the given topic. The topic is covered exhaustively in a top-down way, leveraging links to other CHPs as much as possible. The *Digital Audio Resampling Home Page*[8] is a prototype CHP devoted to sampling-rate conversion. It presently consists of

- A link to a general definition and introduction
- Links to available open-source software
- An in-depth tutorial
- Links to related websites

Concept home pages are under development for other topics which support CCRMA teaching and research. However, the great majority of current links are generated automatically from online document index files. While a cross-document index link can function well as a pointer to supporting material, it is not usually sufficiently comprehensive to be considered a concept home-page. The CHP can provide a much better initial orientation to the topic, unlike a subsection of some remote document. Nevertheless, index links are serving quite well as placeholders for now.

[4] http://www-ccrma.stanford.edu/~jos/od/
[5] http://web.mit.edu/newsoffice/nr/2001/ocw.html
[6] http://mathworld.wolfram.com
[7] http://www.treasure-troves.com
[8] http://www-ccrma.stanford.edu/~jos/resample/

7 Recordings

Recordings of works realized at CCRMA include the following:

- *Arctic Contrasts*, Min Ensemble. Matthew Burtner, "Sikuigvik". Euridice Records (EUCD#012), 2000.
- *II SBCM (II Brazilian Symposium on Computers and Music)*, – digital compact disk containing Celso Aguiar – "Piece of Mind", DISC MFG., INC., BHS1046, Brazil, 1996.
- Matthew Burtner. *Portals of Distortion: Music for Saxophones, Computers, and Stones.* Innova Records (Innova 526), 1999.
- *Cache*, Compilation of winners of the Young and Emerging Sound Artists competition sponsored by the CEC and SOCAN. Michael Gurevich, "Soft White". Canadian Electroacoustic Community, 2000.
- *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.
- Chris Chafe. *Arco Logic*, Solo composer album. Centaur Records, Inc. (CRC2514), 2001.
- Chris Chafe. *Arcology*, Performer/Composer album. Centaur Records, Inc. (CRC2515), 2002.
- Chris Chafe. *Extrasensory Perceptions*, with Greg Niemeyer, 2002.
- *John Chowning,* – Phoné, Turenas, Stria, and Sabelithe. Digital compact disk, WER2012-50 Wergo, Germany, 1988, Harmonia Mundi, Distributors.
- *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.
- *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.
- *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).
- *Computer Music Journal Volume CD,* – digital compact disk to accompany the 20th Anniversary Issue includes Chowning – "Turenas", MIT Press, Cambridge, MA, 1996.
- *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.
- *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.
- Janet Dunbar. *Spirit Journey* Original electroacoustic and computer music with spoken and sung poetry and guitar and synthesizer improvisation. Amberlight Productions (JD10SJ), 1999.

- *Electroacoustic Music II.* Music by Berger, Child, Dashow, Duesenberry, Shapiro (Jonathan Berger: "An Island of Tears"), Neuma 450-73-[J].

- *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.

- 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.

- *Musica Nova 2000 Final Protocol CD.* Matthew Burtner, "Fern", First Prize, Musica Nova 2000 Electroacoustic Music Competition. Society for Electroacoustic Music of Czech Republic, Ministry of Culture, Czech Music Council, 2000.

- *New Music for Orchestra.* VMM 3024, Vienna Modern Masters, 1994. Works by Jaffe ("Whoop For Your Life!") and others.

- *Night Chains.* (Composers Recordings Inc - Emergency Music Series, CRI CD681). Jeffrey Krieger, electronic cello (Jonathan Berger: "The Lead Plates of the Rom Press").

- *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.

- *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").

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.

Ben-Tal, O., Berger, J., and Daniels, M. (2001*a*). De natura sonoris: Sonification of complex data. In *Mathematics and Simulation with Biological, Economical and Musioacoustical Applications; Mathematics and Computers in Science and Engineering*, p. 330. WSEAS Press. Edited by Attellis, Kluv and Mastorakis.

Ben-Tal, O., Berger, J., and Daniels, M. (2001*b*). Sonification in science and art. In *Proceedings of the 2001 ESCOM Conference, Liege, Belgium.*

Berger, J. (1996). Modeling the degree of realized expectation in functional tonal music: A study of perceptual and cognitive modeling using neural networks. In ICMC (1996).

Berger, J. (1998). Connected to what? *Journal of the College Music Society.* Plenary address, 1997.

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