CENTER FOR COMPUTER RESEARCH IN MUSIC AND ACOUSTICS
DEPARTMENT OF MUSIC, STANFORD UNIVERSITY
REPORT NO. STAN-M-106

CCRMA OVERVIEW

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EDITED BY
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Contents

1 General Information

CENTER FOR COMPUTER RESEARCH IN MUSIC AND ACOUSTICS
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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 with an annual outdoor computer music festival in July. In-house technical reports and recordings are available, and public demonstrations of ongoing work at CCRMA are held periodically.

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

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

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

2 Roster

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

2.1 Staff and Faculty

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

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

2.2 Music Graduate Students

Login	Name	Degree Program
cburns	Christopher Burns	PhD Computer-Based Music Theory and Acoustics
lonny	Lonny Chu	PhD Computer-Based Music Theory and Acoustics
pchordia	Parag Chordia	PhD Computer-Based Music Theory and Acoustics
pph	Patty Huang	PhD Computer-Based Music Theory and Acoustics
tkunze	Tobias Kunze	PhD Computer-Based Music Theory and Acoustics (on leave)
randal	Randal Leistikow	PhD Computer-Based Music Theory and Acoustics
unjung	Unjung Nam	PhD Computer-Based Music Theory and Acoustics
cnichols	Charles Nichols	PhD Computer-Based Music Theory and Acoustics
norton	Jonathan Norton	PhD Computer-Based Music Theory and Acoustics
sile	Sile O'Modhrain	PhD Computer-Based Music Theory and Acoustics
juan	Juan Pampin	PhD Computer-Based Music Theory and Acoustics (on leave)
craig	Craig Sapp	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
savd	Scott Van Duyne	PhD Computer-Based Music Theory and Acoustics (on leave)
rswilson	Scott Wilson	PhD Computer-Based Music Theory and Acoustics
leigh	Leigh VanHandel	PhD CCRMA-cology
aguiar	Celso Aguiar	DMA Composition
oded	Oded Ben-Tal	DMA Composition
mburtner	Matthew Burtner	DMA Composition
ching	Ching-Wen Chao	DMA Composition
falk	Chris Falk	DMA Composition
rjfleck	Robert Fleck	DMA Composition
cwjones	Christopher Jones	DMA Composition
dkeller	Damián Keller	DMA Composition
senylee	Seungyon-Seny Lee	DMA Composition
bobby	Bobby Lombardi	DMA Composition
jugend	Jugend Nurit	DMA Composition
cprestia	Chrysa Prestia	DMA Composition
kotoka	Kotoka Suzuki	DMA Composition
be	Brook Eaton	MA Science and Technology
hipple	Aaron Hipple	MA Science and Technology
kikuno	Kimberley Johnsen	MA Science and Technology
jleboeuf	Jay LeBoeuf	MA Science and Technology
anrew	Andrew Roper	MA Science and Technology (on leave)
issac	Issac Roth	MA Science and Technology (on leave)

2.3 Engineering Graduate Students

Login	Name	Degree Program
bilbao	Stephan Bilbao	PhD Electrical Engineering
bacook	Bryan Cook	PhD Electrical Engineering
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
vickylu	Hui-Ling Lu	PhD Electrical Engineering
purswel	Elizabeth Purswell	PhD Electrical Engineering
putnam	William Putnam	PhD Electrical Engineering (on leave)
stilti	Timothy Stilson	PhD Electrical Engineering (on leave)
harv23	Harvey Thornburg	PhD Electrical Engineering
traube	Caroline Traube	PhD Electrical Engineering
yoonie	Yoon Kim	PhD Electrical Engineering

2.4 Undergraduate Students

Login	Name	Degree Program
jbyron	Jeffrey Byron	Music, Science and Technology
dkc	David Chisholm	Music, Science and Technology
n/a	Kim Gaston, Jr.	Music, Science and Technology
cgignoux	Christopher Gignoux	Music, Science and Technology
sandyg	Sandy Greenfield	Music, Science and Technology
n/a	John Horton	Music, Science and Technology
n/a	Anthony Kalomas	Music, Science and Technology
twist	Colin Matheson	Music, Science and Technology
jquiroz	Jennifer Quiroz	Music, Science and Technology
requenez	Edward Requenez	Music, Science and Technology
croyer	Christiaan Royer	Music, Science and Technology
mbatstan	Michael Bautista	Music, Science and Technology (minor)
n/a	David Byron	Music, Science and Technology (minor)
mcc6	Miguel Chavira	Music, Science and Technology (minor)
thefisch	Jeffrey Fischer	Music, Science and Technology (minor)
ericm	Frederick Miller	Music, Science and Technology (minor)
hangwire	William Slater	Music, Science and Technology (minor)
n/a	Ned Tozun	Music, Science and Technology (minor)

2.5 Visiting Scholars

Login	Name	Home Affiliation	Term
bau	Marcia Bauman	Lecturer. Music Department	ongoing
jdc	Joanne Carey	Composer. USA	ongoing
jan	Jan Chomyszyn	Psychoacoustic Research. Poland	ongoing
mo	Maureen Chowning	Singer/Composer. USA	ongoing
n/a	Dr. Baoqiang Han	China	6/2000 - 12/2000
herbst	Christian Herbst	Research. Austria	9/99 - 4/2000
daj	David Jaffe	Composer. Software Research Engineer. USA	ongoing
peer	Peer Landa	Composer. Norway	ongoing
levitin	Daniel Levitin	Psychology Research/Lecturer. USA	ongoing
n/a	Stephen Lew	England	7/2000 - 9/2000
eig	Enrique Moreno	Research. Mexico	ongoing
muller	Carl Muller	Research. USA	ongoing
jcpark	Jong-Chul Park	Prof. of Composition. Korea	9/98 - 8/2000
pasi	Fiammetta Pasi	Composer. Italy	ongoing
juanig	Juan Reyes	Research. Columbia	9/99 - 9/2000
roebel	Axel Roebel	Germany	4/2000 - 10/2000
patte	Patte Wood	ICMA. USA	ongoing
n/a	Salvi Ystad	Research. France	6/2000 - 6/2001

2.6 Collaborators

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

2.7 Industrial Affiliates

Company	Address
Digidesign	Palo Alto. CA
Hewlett Packard	Palo Alto. CA
Interval Research Corporation	Palo Alto. CA
NTT Basic Research Labs	Kanagawa. Japan
Opcode Systems	Palo Alto. CA
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 fast Pentium class PCs running Linux (some of them dual-booting Linux and NEXTSTEP), 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 network. Soundfile manipulation and MIDI input and output are supported on all platforms. Multichannel playback is supported on some Linux and SGI workstations and on the Macs through several Pro Tools systems. 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, Singular Solutions analog and digital audio input systems for the NeXTs, and several Panasonic DAT recorders. Text and graphics are handled by an HP 4c color scanner on the unix-based systems and by high resolution network connected printers.

The recording studio consists of a control room and an adjoining recording studio. Equipment available currently includes two Tascam DTRS 8-track digital recorders (DA-78HR and DA-38), 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 Lexicon 224XL digital reverberator, Westlake BBSM-10 and JBL 4206 monitors, and outboard gear including equalizers, compressors, and digital effects processors. A Macintosh-based sequencer playback system and a Linux PC-based computer system are available in the control room. 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 the MIDI studio is organized around a PowerMac 8600 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 MIDI Studio audio system includes a Mackie 24-8 analog mixer, Tascam DA-38, Panasonic SV-3700 DAT recorder, Denon DN-600F CD player, and ProTools—24 with 888 I/O. Monitoring is via four Ramsa WS-A200 speakers with Yamaha P-2200 power amps. 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 4 or Peak programs.

Studio E is a ProTools III-based room with some MIDI capability. Audio equipment includes a Tascam DA-88 w/ IF-88AE AES/EBU converter, Allen and Heath GL2 mixer, and Genelec 1030A monitors. The ProTools system features an 888 I/O module and NuBus expansion chassis with nine DSP Farm cards for a powerful mixing environment. Several ProTools plug-ins are available. MIDI equipment includes an E-Mu Emulator IV, Korg X3R, and a Kurzweil K2000 keyboard. A Linux system is available with Sonorus audio card providing 8-channel digital I/O.

Studio D is CCRMA's digital editing and dubbing facility. Equipment available includes a Studer-Editech Dyaxis II digital editing processor, a PowerMac 7100, an SGI computer with 8-channel digital connection to a Tascam DA-88, a Panasonic SV-3700 DAT recorder, a Denon digital-output CD player, a Sony PCM-601 PCM processor with serial digital output, a Yamaha DMP7D digital mixer, a Yamaha SPX-1000 digital effects processor, a Tascam 122 mk II cassette recorder and monitor speakers by Meyer Sound Labs (Model 833) for stereo and Boston Acoustics A-60 speakers for quad monitoring.

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 are the MA in Music, Science, and Technology; the DMA in Composition; and the PhD in Computer-Based Music Theory and Acoustics. Acceptance in music theory or composition is largely based upon musical criteria, not knowledge of computing. Admission requirements for degree programs can be obtained directly from each particular department. CCRMA does not itself offer a degree.

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

4.1 University Courses at CCRMA

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

Courses offered at CCRMA include:

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

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

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

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

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

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

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

  Survey of the development of music technology. Analysis and aesthetics of electronic music.

- **Music 192. Theory and Practice of Recording**

  - **192A. Foundations of Sound Recording Technology.**

    Topics: elementary electronics, physics of transduction and magnetic recording of sound, acoustic measurement techniques, operation and maintenance of recording equipment, recording engineering principles, microphone selection and placement, grounding and shielding techniques.

  - **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 252. Seminar: Topics in Computer Music.**

  Various topics according to interest.

- **Music 253. Musical Information: An Introduction.**

  Explores the diverse kinds of the musical information used in sound, graphical, and analytical applications. Device-independent concepts and principles in music representation and musical research objectives (repertory analysis, performance analysis, theoretical models, similarity and stylistic simultaion) will be emphasized. Examples will be drawn primarily from Western art music.

- **Music 254. Musical Representation and Computer Analysis: Seminar.**

  Offers an opportunity for participants to explore issues introduced in Music 253 in greater depth and to take initiative for research projects related to a theoretical or methodological issue, a software project, or a significant analytical result.

- **Music 319. Research Seminar on Computational Models of Sound Perception.**

  CCRMA hosts a weekly Hearing Seminar. All areas related to perception are discussed. but the group emphasizes topics that will help us understand how the auditory system works. Speakers are drawn from the group and visitors to the Stanford area. Most attendees are graduate students. faculty. or local researchers interested in psychology. music. engineering. neurophysiology. and linguistics. 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 Signal Processing (DSP) and the Discrete Fourier Transform (DFT).**

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

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

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

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

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

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

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

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

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

4.2 Workshops

CCRMA also offers a series of 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 from the CCRMA WWW Home Page[2]. 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 and the now open source MidiShare environment. Environments for sound synthesis and composition will include the common lisp based clm system, STK (c++), pd (c) and jmax (java/c). Many other interesting tools like the snd sound editor (and its close ties to clm) 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 covers analysis and synthesis of musical signals based on spectral and physical models. It is organized into morning lectures covering theoretical aspects of the models, and afternoon labs. The morning lectures present topics such as Fourier theory, spectrum analysis, the phase vocoder, digital waveguides, digital filter theory, pitch detection, linear predictive coding (LPC), and various other aspects of signal processing of interest in musical applications. The afternoon labs are hands-on sessions using SMS, the Synthesis Toolkit in C++, SynthBuilder, and other software systems and utilities. The lectures and labs are geared to a musical audience with basic experience in math and science. Most of the programs used in the workshop are available to take.

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

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

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

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

- **Interactive Composition and Performance with Computers**

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

  Hands on participation will use the Max programming environment and Common Music, a language that runs on Macintosh, PC and Unix based platforms. It will also involve real-time interaction using the Mathews-Boie Radio Baton (MIDI conductor/controller device). This course is

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

particularly geared towards performers with an interest in interactive performance, improvisation and other ventures into the world of music technology. Emphasis will be on group performance projects, composition of new works, and realizations of existing interactive works.

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

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

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

- **Introduction to Computer-Based Composition**

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

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

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

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

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

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

- **Intensive Audio Digital Signal Processing**

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

5 Compositional Activities

5.1 Overview

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

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

5.2 Composers and Works

Recent compositional works realized at CCRMA include the following:

Oded Ben-Tal

- *Socratic Dialogues* for violin, percussion, and 8 winds (1999)
- *Soliloquy* for cello and computer generated sound (1999)
- *Saraband* for flute, clarinet, violin, cello, piano, percussion, and computer generated sound (2000)

Jonathan Berger

- *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.

- *Marcatto Sempre* (2000) for clarinet (commissioned by Ensemble Kaprizma).
- *Miracles and Mud* (1999) for String Quartet.
- *Of Hammered Gold* (1999) for period instruments and digital bird organ (Chamber Music America Millenium Commission).
- *Arroyo* (1998) for flute, clarinet, piano, motion detected dancer, and computer sound and graphics.

  Commissioned by the Rockefeller Foundation and the Institute Nacional de Bellas Artes, Mexico. Performances: Centro Nacional de Bellas Artes, Mexico City - April 4 1998; Callejon del Ruiz Festival, Guanajuato, Mexico - April 7 1998; Mind Machines and Electronic Culture: The Seventh Biennial Symposium on Arts and Technology, Connecticut College, March 3; Stanford University, May 24th 1999.

- *Elegy* (1998) for alto and chamber ensemble.
- *Con Carne* (1998) for pianist and disklavier.

  Jonathan Berger is a National Endowment for the Arts Fellow, 1996-1997 and recipient of the Chamber Music America Millenium Commission.

Chris Burns

- *Questions and Fissures* for soprano saxophone and CD (1999)

  Questions and Fissures explores the fusion of divergent musical elements. The two loudspeakers present independent voices, kept separate throughout the piece, while the saxophone provides another layer, distinct from the electronic world. Each element pursues its own path of development, corresponding with the others only at the broadest levels of form. In spite of all the ways in which these materials attempt to achieve independence, we hear one piece, and not three – each layer informs and enriches our hearing of the others.

  This piece is the second in a series of works which use speech sounds as both timbral material and organizing forces. The electronic component is composed entirely of heavily processed recordings of my speaking voice. While the speech never quite becomes intelligible, it animates the sound and provides rhythmic structure. In turn, the saxophone part is derived from a rhythmic transcription of the spoken text. Like the speech sounds themselves, the transcribed rhythms never appear intact. Instead, I use them as the basis for a series of variations and distortions.

  Questions and Fissures is dedicated to Matthew Burtner.

- *Strain* for four-channel tape (1999)

  Many of the sounds in Strain are barely audible, the details just beyond reach. Others are noisy, marginal, the kinds of things composers usually work to exclude from their pieces. Perhaps here they find their place.

  Strain is based almost entirely upon recorded speech. I chose to camouflage and obscure this material for a number of reasons - not least because I wasn't willing to listen to recordings of my own voice over and over again while working on the piece. If the texts leave only indirect traces of their presence, they animate the music nevertheless, creating the rhythmic structures and sonorities of the composition.

  Strain uses its four loudspeakers as a quartet of voices, producing a coherent sense of ensemble. An artificial space is not a goal of the piece, and there is no panning or reverberation of any kind. The loudspeakers are in no way "humanized" through this procedure, but I feel that their material presence becomes an explicit feature of the piece.

C. Matthew Burtner

- *Stone Phase* for computer-generated tape (1999)
- *Frames/Falls* for amplified violin, amplified double bass, and electronics (1999)
- *Noisegate 67* for computer metasaxophone (1999/2000)
- *Oceans of Color* for 27 solo saxophones (2000)
- *Signal Ruins* for prepared piano, bass drum, and electronics (2000)

  A new CD, "Portals of Distortion: Music for Saxophones, Computers and Stones" (Innova 526), was released in February 1999. The Wire calls it "some of the most eerily effective electroacoustic music I've heard;" 20th Century Music says "There is a horror and beauty in this music that is most impressive;" The Computer Music Journal writes "Burtner's command of extended sounds of the saxophone is virtuostic...His sax playing blew me away;" and the Outsight Review says "Chilling music created by an alchemy of modern technology and primitive musical sources such as stones...Inspired by the fearsome Alaskan wilderness, Burtner's creations are another example of inventive American composition."

Chris Chafe

- *Transect* for CD (1999)

  99% pure synthesis, *Transect* is another study to create "chamber music" using the current technology. Ingredients of lines, articulations and phrasing were created by playing the synthesizer with a synthetic player whose bow arm loosely mimics the physics of the real thing. A bowed string and a throat were combined for the instrument. A year in the mountains of Alberta and California, and the mid-life interests of the composer figure into the story line, which is like the title, a section traversing.

- *Whirlwind I and II* (1998)

  Violist Ben Simon wondered what it would feel like to be wired into the same computer rig that I developed for my celletto piece, *Push Pull*. He is the first violist to be so inclined and I took that as his consent to be subjected to further devious experimental situations, from which the first version took shape. The result is an antiphonal setting, in which his two violas (one of them electronic) are paired with musical settings from the electronics. The overall setup is a sort of solo version of Ornette Coleman's Prime Time, a duo of quartets in which an acoustic group trades-off with an electronic one.

  The positions of the two violas are tracked by infrared to give the soloist control over events generated on-the-fly by the computer. In creating these materials, I wanted to establish a feeling of vorticity and the approach to vorticity. Hence the name, which incidentally refers to the first real-time digital computer (from the 40's).

A second version for saxophone solo has been played by Maria Luzardo (Arg.) and Katrina Suwalewski (Den.). Its setup and form is closely related to the earlier version.

- *Push Pull* (1995)

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

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

- *Between the Sheets* (1997)

  With Fred Malouf for electric guitar, tenor saxophone, celletto and computers. Premiered ICMC Thessaloniki, Greece, 27 Sep 1997. Also performed in Germany (1998) and U.S. (1999).

  Chris Chafe is an awardee of National Endowment for the Arts Composer's Fellowship 1982–1983, 1994–95; Green Faculty Fellowship 1995-96; Resident Artist, Banff Centre, 1998–99.

Ching-Wen Chao

- *String Quartet No. 2* (1998-1999)

  String Quartet No. 2, inspired by the relationship between soloists and accompaniment in Chinese Opera, explores the idea of two contrasting gestures: a long-sustained note against short, "shattered" figures. The long note is held almost throughout the piece while these shattering sounds try to break up the texture. Additionally, a great deal of "sul ponticello" and harmonics are employed to simulate the high-frequency, nasal singing of the soloists.

  The pitch A provides a focal point to the piece. It presents itself both in long, sustained gestures and it also forms the background of the harmonic workings of the piece.

  In 1999, *String Quartet No. 2* won the first prize of the Young Composer Competition in the annual ACL (Asian Composer League) conference, and the first prize in Music Taipei 1999, the most prestigious composition competition in Taiwan.

- *Studies for 2 Pianos* (1999)

  To compose a series of studies for 2 pianos has been in my compositional plans for some time. The idea is to employ the serial manipulation of pitch, rhythm, dynamics, timbre, new piano techniques, etc., to achieve less predictable results.

  Study I explores the idea of two contrasting entities: long and loud notes (foreground) against short and soft ones (background). Midway through the piece, the 2 roles seem to exchange. (The 54-note series overwhelms the piece pitchwise, and a series of prime numbers, 1, 3, 5, 7, 9, 11 and 13, decides the number of rapid notes for the succession of each phrase.)

  Study II presents accented notes in extremely fast ascending scales between the 2 pianos and a slow descent.

  Study III, while the third in this series, also belongs to a series of pieces dedicated to the memory of my father. As in all these dedicatory compositions, the pitches G# and C (his initials) are highlighted.

- *Soundstates* (1998) for percussion and tape

  Soundstates presents and 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 coda carries residual traces of preceding elements. The source sounds were mostly drawn from the marimba, played by Randal Leistikow. They were digitally processed in the CLM (Common Lisp Music) environment. Many thanks to Juan Pampin who helped me in employing CLM equipment, and to Randal's performance.

  Soundstates was premiered at CCRMA in Fall 1998 and was recently performed at the International Computer Music Conference, October 1999.

- *Counterattack* (1999) for clarinet and delay

  Delay lines, as "counterattack", begin by echoing only the strong notes played by the clarinet (processed through an amplitude follower) but gradually take over the performance from the clarinet during the course of five stages. The delay lines utilize various controls of delay time, feedback amount, detectable values, and pitch shifts. the clarinet sound is processed in real-time in the Max/MSP environment.

David A. Jaffe

- *OTHER WORLDS: An homage to Carl Sagan*

  A concerto for Zeta electric/MIDI violin and symphonic band

  Carl Sagan challenged and inspired a generation to consider a universe not made for us, to look beyond our planet, and at the same time to recognize its fragility and preciousness. He played a leading role in space exploration, planetary science, the study of the origins of life and the hunt for radio signals from extra-terrestrial civilizations. I attended a series of lectures by Sagan at Cornell University in the early 70s and have been a fan ever since. In *Other Worlds*, I have tried to paint in sound a vista such as might be seen by the shores of the nitrogen lakes of Triton, freshly covered with methane snow and irradiated into the material of life.

  Other Worlds was commissiond by the 1998 International Computer Music Conference and the University of Michigan, and premiered at the conference. Andrew Jennings was the violin soloist. H. Robert Reynolds was the conductor of the University of Michigan symphonic band. The piece was also presented at the 1999 SEAMUS conference in San Jose.

  BIOGRAPHY: David A. Jaffe received a National Endowment for the Arts Collaborative Fellowship in 1993-1994, an NEA Composer Fellowship in 1989 and an NEA Composer Fellowship in 1982. He was the NEA Composer-In-Residence with Chanticleer 1991. In 1998, his *Other Worlds* was commissioned and premiered at the International Computer Music Conference in Ann Arbor. Other 1997-1998 premieres include *Quiet Places* for string quartet, presented on the San Francisco Symphony Chamber Music series at Davies Hall, *Havana Dreams* for chamber ensemble, presented by Earplay at Yerba Buena Forum, and *The Seven Wonders of the Ancient World*, presented at Yerba Buena Theatre.

Christopher Wendell Jones

- *Matragn* for clarinet and computer generated sound (1999)

  The title of this piece reflects its structural nature. *Matragn* is an anagram of the word "Tangram," a Chinese puzzle. The puzzle is solved by arranging seven simple shapes in a multitude of configurations to create new, more complex forms. Like the puzzle, Matragn consists of simple elements which are perpetually reordered and reinvented.

Matragn was written for clarinetist/composer/improviser Matt Ingalls, whose improvisations provided the sonic core for the electronic part. Special thanks also to Chris Burns and Juan Pampin for their technical advice and support.

Damián Keller

- *touch'n'go / toco y me voy* for eight-channel tape and actor (1998-1999)

  touch'n'go / toco y me voy is a modular text and music work. Each section is a complete, self-contained piece which shares material and sound references with the other sections. The piece is published as an enhanced CD by earsay productions http://www.earsay.com. The CD includes sounds, text, and the algorithms used to create the piece. The text, in HTML format, is available at http://www.sfu.ca/~dkeller.

  Sections of *touch'n'go* have been played in Vancouver, Bourges, Stanford, and on Zagreb Radio.

- Current Projects:
  - *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.

Jun Kim

- *Eum-Yang* (1998)

  Eum-Yang is a composition for Disklavier, sampled and computer-modified violin sounds, and Celleto. The Disklavier and violin sounds are controlled by Radio-Baton through the PadMaster program using a NeXT computer. Two digital mixing processors (DMP-11) are also linked to the Radio-Baton to control the quadraphonic sound system.

  Eum-Yang, in chinese pronunciation Yin-Yang, is an old oriental philosophy. "Eum" means dark and cold, while "Yang" means bright and hot. In music, these constrasts and polarity can be expressed in many ways: Color of harmony (dark and bright), Level of pitches (low and high), Level of loudness (soft and loud), and speed of rhythm (fast and slow).

  The symbol of Eum-Yang, called Taeguk, is divided into two Yee (Divine Gender), which are in turn divided into four Sang (Divine Phase). The four Sang are divided into eight Kweh (Divine Diagram). Each of these eight Kewh has a meaningful names which are four polaric pairs: Sky and Earth, Pond and Mountain, Fire and Water, and Thunder and Wind. The piece contains twelve sections which are eight sections of each of above and four sections of each of those four pairs, which is a kind of recapitulation.

Seungyon-Seny Lee

- *Je est un Autre* (Self is Other) (1999/2000)

  Je est un Autre is a journey of imagination and an aesthetic evolution of its ingredients. The total work will consist of four 5-10 minute pieces: combining electro-acoustic music, and visual images in the first piece, and performance elements in the second piece. The third piece will consist of only acoustic instruments, a sextet, and the fourth piece will be performed with a ceramics exhibition in a gallery in Rome, Italy.

  In these 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 raw acoustic material will be recordings of the sound of running water, shaking rice, and insects. The sounds will be electronically processed, creating a broad palette of timbres and sonic textures. These transformations will be used to develop diverse musical layers, inviting the listener to interpret the music through his own imagination.

The imagery for the piece will include animation and photographs. The images were chosen for their symbolic reference to internal psychological states. The animation will be an abstraction of the photographs, signifying the elements of the unconscious which form wishes, desires, and symbols.

In *Je est un Autre I*, a fish tank will be 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.

In addition to images and sound, *Je est un Autre II* will incorporate a performance artist. This piece will continue to explore concepts presented in *Je est un Autre I* with the performer personifying specific facets of the unconscious. The actions of the performer will illuminate three phases of the unconscious: the Real (need), the Imaginary (demand), and the Symbolic (desire) using a plastic installation.

Premieres:

- *Je est un Autre I*: Stanford, California (Dec. 1999)
- *Je est un Autre III*: Stanford, California (May 2000)
- *Je est un Autre IV*: Stanford, California (Aug 2000)

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.

- *House of Mirrors*

  > "...come, travel with me through the House of Mirrors, the one outside me and the one within. Run, fly, never stop ... never think about being lost in the maze of illusions, or you will be. Glide with me through rooms, doors and corridors, surfing on tides of time, looking for that universe left behind an eternity ago. Listen to the distorted steps, the shimmering vibrations that reflect in the darkness, watch out for the rooms with clocks where time withers and stops ..." fll.

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

Charles Nichols

- *Interpose* for guitar and computer generated tape (1995/1999)

  Interpose is a study in the interposition of gestural content on the local and structural level. Materials are introduced on their own and then incorporated into the overall texture, or taken from the texture and elaborated upon within their own sections.

  The pitch content is taken from rotations and transpositions of a row built from trichords and tetrachords, which themselves are the basis for the harmonic motion of the piece. The row also serves as a skeletal pitch structure for the piece, providing the pitch levels for each section.

  The tape part serves as a timbral extension of the guitar part, as if the resonance of the guitar is being transformed. The timbres of the tape part were created with instruments written in Common Lisp Music which use a hybrid approach to additive synthesis.

  Building on the long tradition of additive synthesis, various conventional synthesis techniques are used to resynthesize the individual partials of an analyzed sound, which are added to produce the resynthesized sound. The frequencies and amplitudes of the individual partials of an analyzed sound are converted to percentages of the fundamental frequency. Then the frequencies and amplitudes of various types of unit generators are set to these values and added to create a spectrum related to the original sound source, but exhibiting the distinct characteristics of the chosen synthesis technique. In addition to sine wave resynthesis, frequency modulation, formant filtered pulse train subtractive synthesis, and Karplus-Strong plucked string physical modeling instruments are used to generate each partial of the resynthesized sound, producing a pure steady-state, spread, scattered, and plucked timbre, respectively. Furthermore, the frequency, amplitude, panning, distance, and reverb of each synthesis instrument are controlled by two envelopes: one which dictates the global behavior for each musical parameter and another which determines the scaling of the global behavior over the range of partials, providing global control over the musical behavior for each partial of the resynthesized sound.

  Performances:

  - Magnus Andersson performed *Interpose* for guitar and computer generated tape at June In Buffalo, 6/99.
  - Cem Duruoz performed *Interpose* at The College Music Society Central Pacific Chapter Annual Meeting, 2/00.
  - Recording of *Interpose* played at VIII Festival Internacional de Musica Electroacustica "Primavera en la Habana 2000" in Havana, Cuba, 3/00.

- *Regulate Six* for MIDI violin and computer (1998)

  Regulate Six is a study in granular synthesis. Samples were taken from recordings of male and female voices singing a line from a children's book, and were reassembled using Bill Schottstaedt's Common Lisp Music to create a new waveform whose spectrum is based on the selected vowel or consonant content of each word. Within the computer-generated sound files, pitches are grouped according to timbral types and sweep across or converge at points along the stereo field. The MIDI violin triggers an array of samples, which are similar in timbre to the background material, performing real-time granulation on the samples through the use of trills and tremolos. The violin's

MIDI pitch is often harmonized through MAX programming, which is controlled by a foot pedal. The pedal also triggers the start of each sound file.

As an undergraduate violin major, Charles Nichols studied composition with Samuel Adler and Warren Benson, at the Eastman School of Music. After receiving his Bachelor of Music degree, he attended Yale University, where he studied composition with Martin Bresnick, Jacob Druckman, and Lukas Foss, and computer music with Jonathan Berger. Interested in composing and performing interactive computer music, and researching digital synthesis and musical instrument design, he is currently pursuing a Ph.D. at Stanford University's Center for Computer Research in Music and Acoustics (CCRMA), where he has studied composition with Richard Felciano, Jonathan Harvey, and Jean-Claude Risset, and computer music with Chris Chafe.

Juan Carlos Pampin

- *On Space* for percussion sextet and electronics (2000)

> The pleasure of space: This cannot be put into
> words, it is unspoken. Approximately: it is a form of
> experience -the "presence of absence"; exhilarating
> differences between the plane and the cavern, between
> the street and your living-room; the symmetries and
> dissymmetries emphasizing the spatial properties of my
> body: right and left, up and down. Taken to its extreme,
> the pleasure of space leans toward the poetics of the unconscious, to the edge of madness.
> Bernard Tschumi, Architecture and Disjunction

On Space reflects on the texts and ideas of a painter, a writer and an architect that shaped Art over the last century.

In his transcendental book "On the Spiritual in Art" (1910), Wassily Kandinsky wrote:

- 1. The Artist, as a creator, has to express what is of himself.
- 2. The Artist, as a child of his time, has to express what belongs to his time.
- 3. The Artist, in service of Art, must express what belongs to Art.

Kandinsky's ideas, especially those of space and expression, made their way into the piece, embodied as sound trajectories in space that behave as points and lines to plane.

Related to the form of the piece is a text by Borges: *La muerte y la brjula* (1942). Along the pages of this fascinating story, a detective (Erik Lnnrot, "an Auguste Dupin" of detectives) finds his own destiny within an infinite labyrinth that is his own city. A series of mysterious deaths equidistant in time and space are the clues that help him find his own death at Triste-le-Roy (south of a mythical Buenos Aires). The music of *On Space* is deployed in different spaces that are all perspectives of the same urban landscape from the four cardinal points (North, West, East, South). As in the text, the same things are replicated ad infinitum, and the idea that we only need three points to find a fourth becomes obsessive.

Years before designing the folies for La Villette in Paris, Bernard Tschumi wrote in his essay *Questions of Space* (1974):

- 1.0 Is space a material thing in which all material things are to be located?
- 1.1 If space is a material thing, does it have boundaries?
- 1.11 If space has boundaries, is there another space outside those boundaries?
- 1.12 If space does not have boundaries, do things then extend infinitely?

In *On Space*. percussion and electronics are combined to sculpt sound in space. somehow trying to answer these questions in terms of sound poetry. The program for the piece was developed as a dynamic urban design where each section is constructed to show a virtual perspective from different vanishing points.

On Space closes a cycle of pieces that explores the materiality of percussion sound: metal (*Metal Hurlant*. 1996). wood (*Toco Madera*. 1997). and skins (*Skin Heads*. 1998). On Space uses the sound materials created in all these works to shape space as a continuous matter. capable of inflexions and changes.

This piece has been comissioned by "Les Percussions de Strasbourg" and GRAME for the openning of the "Musiques en Scene" festival 2000 in Lyon. France.

- *Skin Heads* for percussion trio and computer generated sounds (1998)

  Skin Heads is for percussion trio and computer generated sounds. Skin heads are flat. usually covering an empty space. just a volume of air. Any resemblance with those that you might cross in the streets of Berlin is mere coincidence. Skin heads resonate. becoming the living body of other instruments. altering their sound or even magnifying their presence. *Skin Heads*. for percussion skins trio and electronics. is based on these resonances (skin percussion instruments). explored and transformed both by electronic and acoustic means. *Skin Heads* is the third piece of a cycle written for each family of percussion instruments and electronics. The first two works of the cycle are *Metal Hurlant* (1996). for metallic percussion (solo). and *Toco Madera* (1997) for wooden percussion (two players). both premiered at Stanford. This cycle will be completed with a percussion quartet combining all the instrumental palette.

  Technical note: The spectral analysis and transformations of the sampled percussion instruments were done using ATS. spectral modeling software progrmmed by me in Lisp. All the digital sound processing and synthesis for the piece was performed with Common Lisp Music. developed at CCRMA by Bill Schottstaedt.

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

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

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

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

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

  Born in Buenos Aires. Argentina. Juan Pampin has studied composition with Oscar Edelstein and Francisco Kropfl. He holds a Master in Computer Music from the Conservatoire Nationale Superieur de Musique de Lyon. where he studied with Denis Lorrain and Philippe Manoury. As a Visiting Composer at CCRMA in 1994. he composed the tape piece "Apocalypse was postponed due to lack of interest" that received an award in the Concours International de Musique Electroacoustique de Bourges 1995. He has been composer in residence at the LIEM-CDMC studio in Madrid, and guest lecturer at Quilmes National University in Argentina.

- *Metal Hurlant* for metallic percussion and computer generated sounds (1996)

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

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

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

Bob Sturm

- *50 Particles in a Three-Dimensional Harmonic Oscillator: An Experiment in 5 Movements* piece for four-channel tape (1999)

  Premiered May 24, 1999 in Stanford's Dinkelspiel Auditorium, this piece demonstrates the use of my sonification research as applied to algorithmic composition.

Kotoka Suzuki

- *Yoei* (1999) for computer-generated CD, six percussionists, and a dancer (attached to five real-time activated sensors)

  Yoei is a Japanese word, which describes a sound that rings and flutters in the air, resonating in one's ear long after it has been heard. This piece exploits many different acoustic movements to create this effect, with six percussionists and the electronic sound, surrounding the audience in order to complete the spatial environment. The primary goal of this piece, however, is not merely to create sounds, but to combine the world of the visual with that of sound. I have stretched the role of the dancer from merely visual, to both acoustic and visual - creating a live instrumental performer - a dancer who triggers and controls the electronic sounds in real-time using the five electric sensors that are attached to his/her body. All the computer generated sound sources derive from the sounds of the percussion instruments used in this piece, and similarly, the percussion often imitates the sound of the computer-generated sounds of the CD and the dancer. The percussion sounds were manipulated and recorded for the music of the CD and the dance using sound editing programs such as Sound Editor and CLM.

  Kotoka Suzuki received a B.M. degree in composition from Indiana University in 1994 and a D.M.A. degree in composition at Stanford University in 1999 where she studied with Jonathan Harvey and David Soley. She has also been a fellow composer at several festivals including, Domain de Forget, June in Buffalo, Voix Nouvelles Royaumont, and Darmstadt, where she studied with York Hller, Walter Zimmermann, Brian Ferneyhough, and Franco Donatoni. As an active composer, many of my works have been awarded, commissioned and performed in major venues across the globe. including radio and television broadcasts in U.S. and Canada. Last year. she was selected to be a Resident Artist at the Djerassi Resident Artists Program in California and received the Gerald Oshita Fellowship. Most recently. her work for a sextet *Distortion* was broadcasted by the CBC radio station nationwide. and is currently being performed on a Canada Concert Tour by Continuum for their 1999-2000 concert season. She has composed for both acoustic and electronic means. as well as for dance and films. and her latest work *Yoei* for six percussionists. CD. and a dancer with real-time sensors. was premiered at Stanford in July 1999 with a highlight on ZDTV station (cable television dedicated to computers and technology). She is currently working on her new project for CD and video installation. which will be premiered at CNMAT (center for new music and audio technology) in Berkeley this year.

Sean Varah

- *Outside the Box* (1998)

  Outside the Box premiered at the "Made in Canada Festival" in Toronto Canada. performed by New Music Concerts. under Robert Aitken. This work for flute. clarinet. piano. percussion. violin. and cello. was commissioned by the Fromm Foundation at Harvard. and was broadcast live on the CBC radio show "Two New Hours".

- *Borderline*

  Borderline for cello and tape. premiered April 15th. 1998 at the East Cultural Center in Vancouver. Canada. Commissioned and performed by the Canadian cellist. Shauna Rolston. *Borderline* features a lyrical style in the cello contrasted by a diverse electronic tape part. constructed using various analog modelling synthesis programs.

- *Slipping Image*

  Slipping Image for mixed quartet and tape was performed at the 1998 ICMA conference in Ann Arbor Michigan. It was also chosen to be on the 1998 ICMC Compact Disc.

  Sean Varah is currently working on a commission from the CBC for a piece for flute and tape for the Canadian flautist, Robert Cram to be premiered in April, 1999.

  Realaudio recoridngs of all these works are available at http://www-ccrma.stanford.edu/~cello/.

Marek Zoffaj

- *In Principio Erat Verbum* (1998)

  In Principio Erat Verbum (In the Beginning Was the Word). for tape. is an introduction to a work in progress. Its individual parts are based on several statements from the New Testament. The first three sentences from Evangelium by John were used as the initial text and source of inspiration for this introductory movement. The piece is a reflection upon the dialectic relation between concrete (knowledge. experience) and abstract (intuition) meanings of spoken words and their origin. which is also joined with the sacral roots of human beings. The form of this piece reflects the cirlce model of creation of the World that is hidden in the initial text of St John Evengelium. The composition evolved from material which was collected last year at CCRMA. The principal rhythmic structures. as well as some of the individual samples. were recorded using the Korg Wavedrum instrument and a grand piano. All this material was later pro cessed through Bill Schottstaedt's CLM. Paul Lansky's RT. and ProTools.

  Marek Zoffaj is a vising scholar at CCRMA. Stanford University. as an awardee of the Fulbright Foundation. He has been also finishing his Master in Music at Academy of Drama and Music in Bratislava.

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 The CCRMA Music Kit and DSP Tools Distribution

David Jaffe and **Julius Smith**

New releases (V5.0+) are now made by Leigh Smith of tomandandy and Stephen Brandon at the University of Glasgow, who are porting the Music Kit to OPENSTEP, Apple's MacOsX and MacOsX-Server, Windows98, and Linux. Latest releases and progress can be found at `http://www.tomandandy.com/-MusicKit`.

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

Release 4.2 is an incremental release with several significant additions:

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

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

- UPGRADED INTEL-BASED HARDWARE SUPPORT

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

- MOST RECENT APPLICATIONS

  Two Music Kit applications of note are available separately:

  - *Sequence*, a Music Kit Sequencer developed by Pinnacle Research. The new Sequence 0.9.85 release is available free from the CCRMA ftp server (`ftp://ccrma-ftp.stanford.edu/pub/-NeXT/Sequence.9.85.tar.gz`). This is an updated version released in 1998.
  - *SynthBuilder*, a synthesis instrument design and performance tool. SynthBuilder was the Grand Prize winner of the Second Annual International Music Software Competition at Bourges. It was developed by Stanford University's Sondius program, and is now being

supported and further developed by Staccato Systems Inc. The NEXTSTEP version, including a free license authorization code, is available from http://www.StaccatoSys.com or ftp://ftp.StaccatoSys.com. Staccato Systems is also porting SynthBuilder to Windows 95, using the host CPU to do synthesis.

- RECENT UNIT GENERATORS

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

- OTHER RECENT FEATURES

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

Other Music Kit News

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

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

Music Kit Background

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

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

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

6.1.2 Samply Great

Christian Herbst

Samply Great, a standalone Windows application with a user-friendly graphic interface, is a track-based Sampling/Mixing programme with DSP features. Basic concepts of computer music, such as additive, subtractive and granular synthesis can be explored in a WYSIWYG manner.

The programme uses sound samples, envelopes for additive synthesis (which can be derived from the analysis of an existing sound), and noise as sound sources. Several effects, for instance volume changes, waveshaping or transposition can be applied to the whole score or each track, and also to each note of a track. The effects, as well as the sources, can be varied dynamically over the range of the score and/or each note.

All parameter curves/envelopes can be drawn with the mouse, providing an extremely intuitive working environment. If the computational load is not too great, the output can be heard in realtime (using the Windows Direct Sound API). An output file (WAVE format) is additionally created during each rendering process. The projects can be saved and loaded to and from disk. The option of exporting the whole project as ANSI C code provides the possibility of porting and compiling the project on a platform other than Windows, as well allowing post-processing and fine-tuning of the project.

More information, the executable, and the source code of the C++ library used to create the application will be available online by May 2000 at `http://www-ccrma.stanford.edu/~herbst/samply_great`.

6.1.3 Singsing

Christian Herbst

Voice teachers/pedagogues usually lack an in-depth understanding of the concepts used to analyze the singing voice, a fact which is a considerable obstacle to efficiently putting them into practice. Singsing, a Windows application with a simple graphical user interface, provides basic tools to introduce a nevertheless profound analysis of the singing voice into the process of teaching.

For pitch detection and calculation of the residual signal, Singsing uses the programme Praat and its shell script (as developed by Paul Boersma - `http://www.fon.hum.uva.nl/praat`) as an underlying process. The programme offers the following features: Plots of Pitch Tier, Second Order Perturbation, average wavecycle and error signal, and time-varying spectral plots, as well as spectrogrammes of the input, the residual and the vibrato tier. To be developed is an estimation of the vocal track shape.

The analysis results of each sound file are automatically written or appended to an ASCII output file, which can then be imported into other applications to calculate statistics.

More information and a windows executable file will be available online by late Summer 2000 at `http://www-ccrma.stanford.edu/~herbst/singsing`.

6.1.4 Mi_D

Tobias Kunze

Mi_D is a multi-platform shared library that offers clients a simple and unified, yet unique set of MIDI services not commonly found in existing driver interfaces. Its main design goal was to allow clients to add sophisticated MIDI support to their applications at minimal cost.

See also the Mi_D Home Page at: `http://ccrma-www.stanford.edu/CCRMA/Software/mi_d/doc/`

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

Fernando Lopez Lezcano

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

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

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

6.1.6 A Dynamic Spatial Sound Movement Toolkit

Fernando Lopez Lezcano

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

6.1.7 *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/`

6.1.8 ATS (Analysis/Transformation/Synthesis): a Lisp environment for Spectral Modeling

Juan Pampin

ATS is a library of Lisp functions for spectral **A**nalysis, **T**ransformation, and **S**ynthesis of sounds. The Analysis section of ATS implements different partial tracking algorithms. This allows the user to decide which strategy is the best suited for a particular sound to be analyzed. Analysis data is stored as a Lisp abstraction called "sound". A sound in ATS is a symbolic object representing a spectral model that can be sculpted using a wide variety of transformation functions. ATS sounds can be synthesized using different target algorithms, including additive, subtractive, granular, and hybrid synthesis techniques. The synthesis engine of ATS is implemented using the CLM (Common Lisp Music) synthesis and sound processing language, and runs in real-time in many different platforms. ATS together with CLM provide an environment for sound design and composition that allows the user to explore the possibilities of Spectral Modeling in a very flexible way. The use of a high level language like Lisp presents the advantage of a symbolic representation of spectral qualities. For instance, high level traits of a sound, such as global spectral envelopes, frequency centroids, formants, vibrato patterns, etc., can be treated as symbolic objects and used to create abstract sound structures called "spectral classes". In a higher layer of abstraction. the concept of spectral class is used to implement predicates and procedures. conforming spectral logic operators. In terms of this this logic. sound morphing becomes a "union" (a dynamic interchange of features) of spectral clases that generates a particular hybrid sound instance.

For more information about ATS see `http://www-ccrma.stanford.edu/~juan/ATS.html`.

6.1.9 Spectral User Interface (SUI): real-time spectral transformations in ATS

Juan Pampin

Spectral transformations had became an important tool for electronic music composers in the last few years. While working with spectral models composers usually want to evaluate how wide a range of new sounds is available by spectral transformations of a particular source. Usually these kind of explorations have to be done step by step out of real-time due to their complexity. limiting the composer to a gradual approximation to the results. This kind of approach tends to constrain the composer's ability to combine transformations and to explore different regions of the spectral structure. finally limiting his creative work in this domain. ATS provides a Spectral User Interface (SUI) for real-time spectral transformations. Using real-time CLM capabilities. the SUI provides the user with a set of sliders that control different transformation parameters during resynthesis. In its present version the SUI provides the following spectral controllers:

- AmpScale: amplitude scaling of the spectral components
- Transposition: frequency transposition. All partials of the sound are transposed in frequency as in a variable-speed tape recorder without changing the length of the source (the default range allows transpositions within two octaves. it goes from 0.5 to 2.0)
- Shift: frequency shifting of the partials. This procedure adds a fixed amount of frequency to all the partials of the sound. creating a spectral translation or compression. If we consider a harmonic spectrum generated by the formula y=a*x. where y is the frequency of a partial. x its rank and a the frequency value of the fundamental. the spectral shift can be expressed with the following equation: y=a*x+b. where b is the shift factor. The user controls the amount of shift in terms of a percentage of the fundamental frequency of the sound (the default range goes from 0% to 100%)
- Distortion: this transformation considers that the source has a harmonic structure (linear spectrum) and let the user exponentially distort it. If we consider a harmonic spectrum generated by the formula $y = ax$. where y is the frequency of a partial. x its rank and a the frequency value of the fundamental. spectral distortion can be expressed with the following equation: $y = ax^b$. where b is the distortion factor. If the value of b is 1.0 we obtain a harmonic structure. if we increase its value we get a non linear frequency structure that is perceived as inharmonic (by default the user can adjust the distortion factor within a range of 0.0 to 1.0)
- TimeScale: this slider acts as a time-frame "scrubber". The user can move across frames of the spectral structure during synthesis and even freeze the synthesis at a given frame. Using a toggle button the SUI can be set into "scrubbing" mode or into a loop synthesis mode.

Conclusions: Using ATS's SUI the composer can explore many ways of transforming spectral data during resynthesis. Transformations can not only be dynamic but can also be limited to a particular region of the spectrum by means of the TimeScale slider. Transformations can be compounded to create complex spectral results that the user can explore in real-time. On SGI platforms sliders can be controlled through MIDI so the user can use more ergonomic controllers (like fader boxes. wheels. etc.) to synchronically control several sliders.

6.1.10 Stanford Computer-Music Packages for Mathematica

Craig Stuart Sapp

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

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

6.1.11 The Synthesis ToolKit (STK)

Perry R. Cook and Gary P. Scavone

STK is a set of audio signal processing C++ classes and instruments for music synthesis. You can use these classes to create programs which make cool sounds using a variety of synthesis techniques. This is not a terribly novel concept, except that STK is very portable (it's mostly platform-independent C and C++ code) AND it's completely user-extensible. So, the code you write using STK actually has some chance of working in another 5-10 years. STK currently works on SGI (Irix), Linux, NeXTStep, and Windows computer platforms. Oh, and it's free for non-commercial use. The only parts of STK that are platform-dependent concern real-time sound and MIDI input and output ... but we've taken care of that for you. The interface for MIDI input and the simple Tcl/Tk graphical user interfaces (GUIs) provided is the same, so it's easy to voice and experiment in real time using either the GUIs or MIDI.

STK isn't one particular program. Rather, STK is a set of C++ classes that you can use to create your own programs. We've provided a few example applications that demonstrate some of the ways that you could use these classes. But 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" GUI, you probably don't want to use STK. Spending hundreds of hours making platform-dependent GUI code would go against one of the fundamental design goals of STK - platform independence. STK can generate simultaneous .snd, .wav, and .mat output soundfile formats (beside realtime sound output), so you can view your results using one of the numerous sound/signal analysis tools already available over the WWW (e.g. Snd, Cool Edit, Matlab). For those instances where a simple GUI with sliders and buttons is helpful, we use Tcl/Tk (which is freely distributed for all the STK supported platforms). A number of Tcl/Tk GUI scripts are distributed with the STK release.

Perry Cook began developing a pre-cursor to 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++, SGIs, added realtime 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/CCRMA/Software/STK/`.

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6.1.12 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-4.tar.gz.

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

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

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

6.1.14 Common Music

Heinrich Taube

What is Common Music?

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

History

Common Music began in 1989 as a response to the proliferation of different audio hardware, software and computers that resulted from the introduction of low cost processors. As choices increased it became

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

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

Implementation

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

Synthesis Control

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

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

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

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

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

Contact

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

6.2 Physical Modeling and Digital Signal Processing

6.2.1 Synthesis of Transients in Classical Guitar Sounds

Cem Duruoz

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

6.2.2 Estimation of Multiple Fundamental Frequencies in Audio Signals Using a Genetic Algorithm

Guillermo Garcia
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A method for estimating multiple, simultaneous fundamental frequencies in a polyphonic audio spectrum is presented. The method takes advantage of the power of genetic algorithms to explore a large search space, and to find a globally optimal combination of fundamental frequencies that best models the polyphonic signal spectrum. A genetic algorithm with variable chromosome length, a special crossover operator and other features is proposed. No a-priori knowledge about the number of fundamental frequencies present in the spectrum is assumed. Assessment of the first version of this method has shown correct detection (in number and value) of up to five fundamental frequencies. Planned refinements on the genetic algorithm operators could enhance this performance.

6.2.3 Computation of Reflection Coefficients for an Axisymmetrical Horn by Boundary Element Method

Shyh-Kang Jeng

It seems that there is no literature about using the boundary-element method (BEM) to deal with a music horn, though some authors have applied the BEM to horn loud speaker problems (for examples, Kristiansen, etc. [1], Henwood [2], and Johansen [3]). The BEM approach starts from the Helmholtz equation of linear acoustics, and makes no approximation except those required for numerical calculation. Therefore, it is expected to include the effect of diffraction from edges and the contribution of higher order modes.

In this research, an integral equation is first derived. Special care has to be taken for the singularities. The formulation will take advantage of the axisymmetry, and will express the pressure field inside the cylindrical section as a summation of modal fields. The boundary-element method is then applied to approximate the integral equation by a matrix one. By solving the matrix equation, we may obtain the reflection coefficient directly. Next, the reflection coefficients for a sequence of sampled frequencies in the desired frequency band are computed and an inverse Fourier transform is performed to obtain the impulse response of an equivalent filter. Finally, an approximate FIR or IIR filter is deduced from the equivalent filter, and a physical model of a brass can be obtained by connecting the approximate filter to a digital waveguide system.

With simple extensions, this approach can be used to model bores and openings of wind instruments.

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3 T. F. Johansen. "On the directivity of horn loudspeakers." Journal of Audio Engineering Society, Vol. 42, No. 12, pp. 1008-1019, 1994.

6.2.4 Synthesis of Ecologically-Based Sound Events

Damián Keller and **Chris Rolfe**

We present techniques for the efficient synthesis of everyday sounds, that is, sounds like rain, fire, breaking glass, scraping and bouncing objects, etc. These sounds present dynamic temporal and spectral states that cannot be described by either deterministic or stochastic models alone (Cook, 1997; Roads, 1997). We propose a conceptually simple method for resynthesizing decorrelated, unique sound events using constrained parametric control of stochastic processes.

Granular synthesis has proven to be a convenient and efficient method for stochastic, time-based synthesis (Truax, 1988). To better control spectral details, we extend asynchronous granular synthesis to include phase-correlation, time-dependent overlap, amplitude scaling, and synchronicity between granular streams. We propose a representation of ecologically-based sound events comprising three control levels: micro, meso, and macro. By having a control structure across all three time resolutions we can better manage time-frequency boundary phenomena, thus taking into account windowing and overlap effects, spectral evolutions, and emergent perceptual properties.

Related Article

- Keller, D. "Introduction to the Ecological Approach." *Virtual Sound*, R. Bianchini and A. Cipriani, Eds. Contempo Edizioni, Italy, 1999. (CD-ROM)

6.2.5 Modeling the Sound Event at the Micro Level

Damián Keller, **Jonathan Berger** and **Conrado Silva**

Environmental sounds present a difficult problem for sound modeling because spectral and temporal cues are tightly correlated. These cues interact to produce sound events with complex dynamics. In turn, these complex sounds form large classes which can be defined by statistical measurements. Thus, environmental sounds cannot be handled by traditional deterministic synthesis methods. The objective of this project is to implement algorithmic tools which allow to define sound events by multilevel parameter manipulation.

Micro-level representations of sounds provide a way to control spectral and spatial cues in sound synthesis. Meso-level representations determine the temporal structure of sound events. By integrating these approaches into a coherent data structure we expect to be able to model sound events with complex dynamic evolutions both at a micro and at a meso level. Consequently, these tools will extend the parameter space of ecological models to include spectral and spatial cues.

6.2.6 Toward a High-Quality Singing Synthesizer

Hui-Ling Lu

Naturalness of the sound quality is essential for the singing synthesis. Since 95% in singing is voiced sound, the focus of this research is to improve the naturalness of the vowel tone quality. In this study, we only focus on the non-nasal voiced sound. To trade off between the complexity of the modeling and the analysis procedure to acquire the model parameters, we propose to use the source-filter type synthesis model, based on a simplified human voice production system. The source-filter model decomposes the human voice production system into three linear systems: glottal source, vocal tract and radiation. The radiation is simplified as a differencing filter. The vocal tract filter is assumed all-poled for non-nasal sound. The glottal source and the radiation are then combined as the derivative glottal wave. We shall call it as the glottal excitation.

The effort is then to estimate the vocal tract filter parameter and glottal excitation to mimic the desired singing vowels. The de-convolution of the vocal tract filter and glottal excitation was developed via the convex optimization technique [1]. Through this de-convolution, one could obtain the vocal tract filter parameters and the glottal excitation waveform.

Since the glottal source modeling has shown to be an important factor for improving the naturalness of the speech synthesis. We are investigating the glottal source modeling alternatives for singing voice. Besides the abilities of flexible pitch and volume control, the desired source model is expected to be capable of controlling the voice quality. The voice quality is restricted to the voice source modification ranging from laryngealized (pressed) to normal to breathy phonation. The evaluation will be based on the flexibility of the control and the ability to mimic the original sound recording of sustained vowels.

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6.2.7 Scanned Synthesis, A New Synthesis Technique

Max V. Mathews, Bill Verplank, and Rob Shaw

Developed at the Interval Research Corporation in 1998 and 1999. Scanned Synthesis is a new technique for the synthesis of musical sounds. We believe it will become as important as existing methods such as wave table synthesis, additive synthesis, FM synthesis, and physical modeling. Scanned Synthesis is based on the psychoacoustics of how we hear and appreciate timbres and on our motor control (haptic) abilities to manipulate timbres during live performance. A unique feature of scanned synthesis is its emphasis on the performer's control of timbre.

Scanned synthesis involves a slow dynamic system whose frequencies of vibration are below about 15 hz. 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. Examples include slowly vibrating strings, two dimensional surfaces obeying the wave equation, and a waterbed. We have simulated the string and surface models on a computer. Our waterbed model is purely conceptual.

The ear cannot hear the low frequencies of the dynamic system. To make audible frequencies, the "shape" of the dynamic system, along a closed path, is scanned periodically. The "shape" is converted to a sound wave whose pitch is determined by the speed of the scanning function. Pitch control is completely separate from the dynamic system control. Thus timbre and pitch are independent. This system can be looked upon as a dynamic wave table controlled by the performer.

The psychophysical basis for Scanned Synthesis comes from our knowledge about human auditory perception and human motor control abilities. In the 1960's Risset showed that the spectra of interesting timbres must change with time. We observe that musically interesting change rates are less than about 15 hz which is also the rate humans can move their bodies. We have named these rates Haptic rates.

We have studied Scanned Synthesis chiefly with a finite element model of a generalized string. Cadoz showed the musical importance of finite element models in the 1970s. Our models differ from Cadoz's in our focus on slow (haptic) vibration frequencies. Our finite element models are a collection of masses connected by springs and dampers. They can be analyzed with Newton's laws. We have generalized a traditional string by adding dampers and springs to each mass. All parameters–mass, damping, earth spring strength and string tension–can vary along the string. The performer manipulates the model by pushing or hitting different masses and by manipulating parameters.

We have already synthesized rich and interesting timbres and we have barely started to explore the range of possibilities in our present models. Many other different models can be conceived. We find the prospects exciting.

6.2.8 Perceptual Audio Coding Based on the Sinusoidal Transform

Juan Pampin and **Guillermo Garcia**

In this work, we have explored the possibilities of the sinusoidal model as a frequency-domain representation for perceptual audio coding of various types of audio signals. We have designed a set of techniques for data rate reduction and developed a codec software prototype consisting of three basic blocks:

1. Partial pruning based upon psychoacoustics masking.
2. Smart sinusoidal frame decimation based upon transient detection.
3. Bit allocation based upon psychoacoustics masking, and quantization.

We have evaluated the codec on monophonic musical instruments (harmonic and inharmonic), polyphonic orchestral music, singing voice and speech. Results have been quite satisfying and have shown that the sinusoidal model can be used to achieve interesting compression factors at high quality, for a wide variety of audio signals. In particular, we believe this work shows that the sinusoidal model is not at all limited to monophonic, harmonic signals, when high quality audio compression is the goal.

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

Craig Stuart Sapp

Sig++ is a set of C++ classes intended for use in writing sound generating/filtering programs by direct coding of flowgraphs schematics for signal processing filters as well as for traditional computer-music unit-generator flowgraphs. The paradigm for generating sound is similar to other Music V-style synthesis programs, such as Csound.

An intent of sig++ is to have very portable code. As a result, example programs using the sig++ library have been compiled on several computer configurations: Linux, Windows 95/NT, OpenStep, NextStep, Sun SPARCStations, HP-UX, and SGI IRIX.

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

Future additions to sig++ will be real-time sound input/output in Windows 95/NT and Linux as well as linking control of sound generation to MIDI using *Improv*.

6.2.10 Acoustic Research and Synthesis Models of Woodwind Instruments

Gary P. Scavone

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

Recent 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 being directed toward the development of improved models of woodwind instrument excitation mechanisms. 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.

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6.2.11 Physical Modeling of Bowed Strings: Analysis, Real-time Synthesis and Playability

Stefania Serafin and Julius Smith

Recent work on the field of bowed string has produced a real-time bowed string instrument which, despite its simplicity. is able to reproduce most of the phenomena that appear in real instruments. Our current research consists of improving this model. including also refinements made possible by the improvement of hardware technology and the development of efficient digital signal processing algorithms. In particular. we are modeling string stiffness whose main effect is to disperse the sharp corners that characterize the ideal Helmholtz motion. This dispersion is modeled using allpass filters whose coefficients are obtained by minimizing the L-infinity norm of the error between the internal loop phase and its approximation by this filter cascade.

We are also analyzing the "playability" of the model built. examining in which zones of a multidimensional parameter spaces "good tone" is produced. This study focuses on the influence of the torsional waves and on the shape of the friction curve. The aim is to analyze which elements of bowed string instruments are fundamental for bowed string synthesizer, and which can be neglected, to reduce computational cost. This work is extended by examining the attack portion of a virtual bowed string. Since a player's aim is normally to reach Helmholtz motion as quickly as possible, we analyze the attack parameters in order to determine the parameter combination that allows the oscillating string to achieve Helmholtz motion in the shortest time.

This research is part of the work done by the Strad (Sistema Tempo Reale Archetto Digitale), group made by CCRMA people working on different aspects of bowed string instruments.

The waveguide physical model we have built runs in real-time under different sound synthesis platforms e.g. Max/MSP, the Synthesis Toolkit, Common Lisp Music.

Another part of this research group consists in building controllers to allow composers and performers to play the model. In particular, we are interested in controllers that incorporate force feedback because they allow us to couple the sound and feel of a given bowing gesture. We are currently constructing an actuated bow and running experiments to discover the role played by tactile and kinesthetic feedback in stringed instrument playing.

6.2.12 FFT-Based DSP and Spectral Modeling Synthesis

Julius Smith

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

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

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

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

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

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6.2.13 Digital Waveguide Modeling of Acoustic Systems

Julius Smith

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

In this context, a *waveguide* can be defined as any medium in which wave motion can be characterized by the one-dimensional wave equation. In the lossless case, all solutions can be expressed in terms of left-going and right-going *traveling waves* in the medium. The traveling waves propagate unchanged as long as the wave impedance of the medium is constant. At changes in the wave impedance, a traveling wave partial transmits and partially reflects in an energy conserving manner, a process known as "scattering." The wave impedance is the square root of the "massiness" times the "stiffness" of the medium; that is, it is the geometric mean of the two sources of resistance to motion: the inertial resistance of the medium due to its mass, and the spring-force on the displaced medium due to its elasticity.

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

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

[4] http://www-ccrma.stanford.edu/CCRMA/Courses/420/

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6.2.14 Signal Processing Algorithm Design Stressing Efficiency and Simplicity of Control

Timothy Stilson

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

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

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

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

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

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

6.2.15 Synthesis and Algorithmic Composition Techniques Derived from the Sonification of Particle Systems; And a Resultant Meta-Philosophy for Music and Physics

Bob L. Sturm

de Broglie's hypothesis from Quantum Mechanics (QM) states a particle can behave as either a particle or a wave. Thus a system of particles could become a complex superposition of dynamic waves. Motivated by this the author develops a method for sonification of particle systems in a logical manner. Thinking of sound in terms of an evolving system of particles, potentials, and initial conditions, a unique position is gained. A direct correspondence between sound composition and many-body physics allows ideas from each field to enrich the other, such as using sound to gain a higher comprehension of a phenomenon, or using radioactivity as a compositional device. One application so far explored has been algorithmic composition using a simulated particle system. It has been readily observed that the composer must also become physicist to make effective musical use of these techniques. Paradoxically, the audience need not be versed in physics to visualize and appreciate what they hear–a sign of a successful analogue. But by the very act of uniting physics and music several interesting questions arise, encouraging a possible meta-philosophy of the two. The traditional purposes, meanings, and practices of each, are challenged: and the results are very pertinent to our current techno-culture. Several sound examples will be presented; and if accepted for programming, the first composition made with these techniques: *50 Particles in a Three-Dimensional Harmonic Potential: An Experiment in 5 Movements.*

6.2.16 A Flexible Analysis/Synthesis Method for Transient Phenomena

Harvey Thornburg

Sinusoidal models provide an intuitive, parametric representation for time-varying spectral transformations. However, resynthesis artifacts result to the degree the signal violates assumptions of local stationarity. Common types of transients (or local non-stationary regions) are abrupt changes in spectra, rapid exponentially-decaying modes, and rapid spectral variations (e.g. fast vibrato, chirps, etc.). These phenomena cover a considerably wider framework than that of onset regions in monophonic contexts. Our extended sinusoidal model proceeds with a presegmentation phase followed by region-dependent modeling and resynthesis. In presegmentation, information-theoretic criteria are used to localize abrupt change boundaries, windows are aligned with segment boundaries, then segments are classified as to local stationarity or transience. Locally stationary regions are handled by a sinusoids+noise model. For transients, we adapt parametric models which naturally extend the sinusoids+noise model, such as the time-varying Prony/Kalman model, to mode decay/variation problems. As well as reducing artifacts, extended sinusoids+noise models permit different kinds of processing to be applied to transients, shown to offer the composer considerable flexibility in timestretching-related applications. Finally, we show applications to the single-channel source separation problem and also to that of rhythm-following using a Bayesian framework to handle side information concerning the change boundaries.

6.2.17 Antialiasing for Nonlinearities: Acoustic Modeling and Synthesis Applications

Harvey Thornburg

Nonlinear elements have manifold uses in acoustic modeling, audio synthesis and effects design. Of particular importance is their capacity to control oscillation dynamics in feedback models, and their ability to provide digital systems with a natural overdrive response. Unfortunately, nonlinearities are a major source of aliasing in a digital system. In this paper, alias suppression techniques are introduced which are particularly tailored to preserve response dynamics in acoustic models. To this end, a multi-rate framework for alias suppression is developed along with the concept of an aliasing signal-to-noise ratio (ASNR). Analysis of this framework proceeds as follows: first, relations are established between ASNR vs. computational cost/delay given an estimate of the reconstructed output spectrum; second, techniques are given to estimate this spectrum in the worst case given only a few statistics of the input (amplitude, bandwidth and DC offset). These tools are used to show that "hard" circuit elements (i.e. saturator, rectifier, and other piecewise linear systems found in bowed-string and single-reed instrument models) generate significant ASNR given reasonable computational constraints. To solve this problem, a parameterizable, general-purpose method for constructing monotonic "softening approximations" is developed and demonstrated to greatly suppress aliasing without additional computational expense. The monotonicity requirement is sufficient to preserve response dynamics in a variety of practical cases. Applications to bowed-string modeling and virtual analog filter emulation are discussed.

6.3 Controllers for Computers and Musical Instruments

6.3.1 Updates on the Radio-Baton Program

Max V. Mathews

The radio-baton research this year has focused on midifiles and the radio-baton. The conductor program has been modified to accept pure type 0 midifiles as scores. This requires some way of specifying trigger points in a midifile. Triggers have been encoded into noteon midi commands with keynumbers 0 through 11. These keynumbers are generally not used because the pitches they produce are below the range that can generally be heard as music. As an alternate approach, trigger points can be automatically added to a midifile corresponding to the time signature of the file. For example, a 3/4 time signature will have 3 triggers in each measure; a 4/4 time signature will have 4 triggers per measure. This work has been done by Andrew Einaudi.

The conductor program is currently being extended to accept type 1 midifiles as scores. This requires sorting the events in the various tracks in the type 1 file into time ordered events.

Midifiles are also being used as source material in the radio-baton *Improv* program. For this purpose, a midifile is parsed and read into a structure in the Improv program memory. This allows the Improv program to have easy access to fragments of the file. Thus, it is possible to repeat (loop) sections of the score as many times as desired by some live performace control, or to vary the tempo of the playback either with baton beats or with a knob for continuous tempo control. Several midifiles can be played at the same time, each with a seperate tempo control, but with algorithms to synchronize the files in various ways. For example, in rock music a "solid" percussion track played at a constant tempo can be synchronized with a flexible-tempo solo track by repeating measures in the percussion track as necessary to synchronize with the solo voice.

Many other ways of using midifile material are envisioned for the Improv program. Sequencers and midifiles are a powerful and widely used ways of composing popular music, so we believe their use in the Improv program will be an important addition.

6.3.2 The vBow: A Haptic Musical Controller Human-Computer Interface

Charles Nichols

Previous electronic musical controllers have either added technology to acoustic instruments to translate the expressive qualities of the instrument into digital data, or employed systems of sensors to capture the performance gestures of the player, to provide the computer musician with an expressive electronic musical controller. The advantage of the former case is that the force-feedback of the acoustic instrument, upon which the traditional player depends and his technical training has been based, is retained. However, the translation of an acoustic signal into a digital protocol is prone to error and quantizes the expression of the instrument, limiting the expressive range of the performer. In the latter case, the expression of the performer may be more accurately and fully translated, but the haptic response of the instrument is lost. The vBow, a virtual violin bow musical controller, uses a design based on robotic technology to both accurately capture gestural expression and provide haptic feedback to the performer.

The vBow senses the violinists stroke through the encoder of a servo, driven by a cable attached to a capstan on the shaft of the motor and to either end of the virtual bow. The servo, in turn, engages the bow, driven by the control software which simulates friction and vibration of the string on the bow, according to the velocity of the performer's stroke. The performance data sensed by the encoder and the haptic feedback instructions driving the servo are bussed through parallel data streams to and from a data acquisition card running at a high sampling rate, resulting in minimum sampling error. The result is a first attempt at simulating the complex interaction between the violinist's bow stroke, the acoustic violin's physical response, and the violinist's reaction to the force-feedback of the acoustical system.

6.3.3 Haptic User Interfaces for the Blind

Sile O'Modhrain and Brent Gillespie

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

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

6.3.4 Incorporating Haptic Feedback into Music Controllers

Sile O'Modhrain

This study investigates the role played by haptic (tactile/kinesthetic) feedback in musical instrument playing. Though musicians rely primarily on their sense of hearing to monitor and adjust the sound being produced by their instrument, there exists a second path through which valuable information about the instrument's behavior can be observed - namely the feedback received via the haptic senses, the senses of touch and kinesthesia. A violinist, for example, uses their sensitivity to pressure and vibration to control bow speed. A trombone player can "feel" where the resonant modes of their instrument are by an increase in vibrations fed back to their lips via the mouthpiece.

In our work, we are leveraging off the musician's unconscious use of combined haptic and auditory cues to design music controllers that combine both forms of sensory feedback. We are developing a prototyping environment which allows us to design the "feel" as well as the sound of an instrument. Using a variety of haptic display devices, we can control parameters of physical models running in STK, and use output from these models to generate forces or vibrations which the player can feel. We are currently running a series of studies to assess the utility of such haptic feedback in musical instrument controllers.

6.3.5 *Improv*: Computer/Performer Interaction Programming with MIDI in C++

Craig Stuart Sapp

Improv is an environment for writing programs that enable musician/computer interation using MIDI instruments. There are two components to *Improv*:

- a library of C++ classes for accessing and handling MIDI Input and output from the computer.
- a set of example programs using the library classes that demonstrate programming possibilities. Example programs are categorized by *environment*. Each type of environment is set up for a particular hardwired MIDI I/O configuration. For example, there is an environment for interaction with synthesizers (*synthImprov*), Max Mathews' Radio Batons (*batonImprov*), as well as an interaction environment between computers via MIDI.

The *Improv* environments have been used in two Stanford courses: Introduction to Music Composition and Programming Using MIDI-Based Systems, and Topics in Interactive Computer-Music Performance. Also, the environment was used this past summer (1998) at a Summer Workshop in Germany at ZKM.

The programming library and environments are designed to be portable to different computer operating systems. Currently example programs can be compiled and run in the Windows 95/NT and Linux operating systems with Intel 75 MHz Pentium CPU's or better.

For more information about *Improv*, visit `http://www-ccrma.stanford.edu/~craig/improv/`

6.3.6 Alternative Controllers for Physical Model Development (and Fun!)

Gary P. Scavone

Two special purpose MIDI controllers, the *Holey Controller* and the *Phoney Controller*, have been created using BASIC Stamp II microprocessors by Parallax Inc. The design of these controllers was inspired by recent work of Perry Cook.

- *The Holey Controller*

  The *Holey Controller* is a modified Yamaha WX11 MIDI wind controller that I created for use in playing my digital waveguide woodwind instruments. Using digital waveguide techniques, I developed an efficient model of a woodwind tonehole that accurately simulates all the states of the hole from fully open to closed. I then implemented an eight-hole woodwind model using the Synthesis ToolKit (STK), which allowed me to manipulate the various toneholes in realtime. The problem then became, "How do I control this model?" All currently available MIDI wind controllers output a single MIDI note number for any particular fingering ... no matter how unconventional this fingering may be. Further, these instruments provide no intermediate finger position information for finger states *between* open and closed.

The solution was to take a Yamaha MIDI wind controller (WX11) and place Force Sensing Resistors (FSRs) under each key to determine the key positions. Between the FSRs and the key, a small piece of foam was inserted. In this way, the FSR was driven in its initial highly nonlinear range. Each FSR was connected via ribbon cable to a BASIC Stamp II microcontroller, which was used to determine the key position and output the result in the form of MIDI ControlChange messages. Because I am also using breath pressure MIDI messages from the WX11, I merge the two MIDI channels before inputing the result to my STK instrument. For more information on the *Holey Controller*, see `http://www-ccrma.stanford.edu/~gary/`.

- *The Phoney Controller* (aka "The Air Phone")

  The *Phoney Controller* consists of a telephone housing, four Force Sensing Resistors (FSRs), an Analog Devices ADXL202 two-dimensional accelerometer, a BASIC Stamp II (BSII) microcontroller, a stupid "on-off" switch, and a cool LED! I am essentially using the BSII as a MIDI sequencer. All sounds at this point have to be generated from an external MIDI synthesizer. The FSRs and the 2D accelerometer are used to control various aspects of the sequencer.

  The *Phoney Controller* was built in part to serve as a "one-man band" at my wedding. But mostly, is was built for goofing around and having fun. Given the memory limitations of the BSII, sequences have to be pretty short. That's the challenge ... coming up with simple but interesting patches that someone will enjoy improvising with for hours. For more information on the *Phoney Controller*, see `http://www-ccrma.stanford.edu/~gary/`.

6.4 Psychoacoustics and Cognitive Psychology

6.4.1 Neural Network Models of Musical Cognitive Activities

Jonathan Berger, Daniel Lehmann, and **Dan Gang**

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

1. Schema based theories of cognition.

2. Processes involved in resolving ambiguities and conflicts of schemas and patterns occurring at different structural or hierarchical levels.

3. Dynamic contextualization, that is, how a context is created, adapted, and accepted or rejected as it unfolds in time.

4. Expectational windows - how contexts create both short range and long range predictions. The interaction of short term and long term memory on these processes.

5. The influence of cyclic or metric organizers on pattern extraction and segmentation.

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

6.4.2 Text on Psychoacoustics

Perry Cook

The lectures from CCRMA's Music 151 course, "Psychophysics and Cognitive Psychology for Musicians" are now published as:

- *Music, Cognition, and Computerized Sound: An Introduction to PsychoAcoustics,*" Perry R. Cook, Editor. MIT Press: ISBN: 0262032562, $50.00 Hardback, CDROM included, January 1999.

This introductory text on psychoacoustics, specifically as it relates to music and computerized sound, emerged from a course that has been taught for many years at Stanford University's Center for Computer Research in Music and Acoustics (CCRMA). Organized as a series of 23 lectures for easy teaching, the book is also suitable for self-study by those interested in psychology and music. The lectures cover both basic concepts, and more advanced concepts illuminated by recent research. Further aids for the student and instructor include sound examples on CD, appendixes of laboratory exercises, sample test questions, and thought problems. The contributors, leading researchers in music psychology and computer music, John Chowning, Perry Cook, Brent Gillespie, Dan Levitin, Max Mathews, John Pierce, and Roger Shepard.

6.4.3 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.

6.4.4 Absolute Pitch, Absolute Tempo, Absolute Loudness

Daniel Levitin

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

More specifically, my current research projects include work on:

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

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

6.5 Machine Recognition in Music

6.5.1 Optical Recognition of Printed Music: A New Approach

Walter Hewlett

Recent projects in optical recognition of printed music have tended to give top priority to the extraction of pitch symbols (i.e., noteheads). Noteheads give some information about duration (i.e., they are filled or unfilled), but definitive information also requires the accurate reading of stems, flags, and beams. Symbols for articulation (staccato marks, dynamics, slurs, and so forth) are sometimes ignored if the intended use of the scanned material is in sound applications.

In an effort to create a scanning front-end for the CCARH databases of classical music, which are stored in an attribute-rich format (MuseData) to support notation, sound, and analysis, we have taken the following approach: large objects are identified first. This clarifies contextual properties that may bear on pitch (key signatures, clef changes, octave-transposition signs), duration (beams, stems, and flags), and articulation (slurs, ornaments, et al.). The pitch content of the notehead is the last item to be recognized and completes the representation.

6.5.2 Realtime Chord Recognition of Musical Sound: a System Using Common Lisp Music

Takuya Fujishima

I designed an algorithm to recognize musical chords from the input signal of musical sounds. The keys of the algorithm are:

1. use of "pitch class profile (PCP)", or an intensity map of twelve semitone pitch classes
2. numerical pattern matching between the PCP and built-in "chord-type templates" to determine the most likely root and chord type.

I introduced two major heuristics and some other improvements to Marc Leman's "Simple Auditory Model" so as to make it a practical one.

I implemented the algorithm using Common Lisp Music. The system worked continuously in realtime on an Silicon Graphics O2 workstation and on Intel PC platforms running Linux. It could take the input sound from the audio input, or from a sound file, and could display the recognition results on the fly.

In the experiments, I first used the pure tone and three other timbres from electronic musical instruments to estimate the potential capability of the system. The system could distinguish all of 27 built-in chord types for chord tones played in the above mentioned timbres. Then I input to the system a 50 second audio excerpt from the opening theme of Smetana's *Moldau*. Without the two heuristics, the accuracy remained around 80 percent level. When they were applied, the accuracy rose to 94 percent ... 196 out of 208 guesses that the system made were musically correct. These experiments showed that the system could recognize triadic harmonic events, and to some extent more complex chords such as sevenths and ninths, at the signal level.

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- Fujishima, Takuya. *Realtime Chord Recognition of Musical Sound: a System Using Common Lisp Music*. Proceedings of the 1999 International Computer Music Conference, Beijing, China, pp. 464–467.

6.5.3 Speaker Normalization for Speech Recognition Based On Maximum-Likelihood Linear Transformation

Yoon Kim

In speaker-independent speech recognition, where a pre-trained model is used to recognize speech uttered by an arbitrary speaker, minimizing the effects of speaker-dependent acoustics is crucial. Acoustic mismatches between the test speakers and the statistical model result in considerable degradation of recognition performance. In this work, we apply linear transformation to the cepstral space, which can be viewed as the Fourier dual of the log spectral space. Gaussian mixture is the most popular distribution used for modeling speech segments. If we restrict the feature transformation to be that of *convolution*, which is equivalent to filtering in the log spectrum domain, the resulting normalization matrix exhibits a Toeplitz structure, simplifying the parameter estimation. The problem of finding the optimal matrix coefficients that maximize the likelihood of the utterance with respect to the existing Gaussian-based model then becomes nothing but a constrained least-squares problem, which is convex in nature, yielding a unique optimum. Applying the optimal linear transformation to the test feature space yielded in considerable improvements over the baseline system in frame-based vowel recognition using data from 23 British speakers, and in isolated digit recognition using the TI digits database, consisting of over 300 speakers.

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6.6 Historical Aspects of Computer Music

6.6.1 Impact of MIDI on Electroacoustic Art Music

Alex Lane Igoudin

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

The study was published as the author's doctoral dissertation at CCRMA, Stanford University. It is available in print from CCRMA. Readers interested in evolution of our field over the last two decades are encouraged to acquaint themselves with the full text of work as it presents a legacy of electroacoustic art music in the 1970s, 80s, and 90s: its methodologies, tools, and practices. Each conclusion of the study arrives supported by a wealth of quotes from the interviews as well as statistic data calculated upon the survey completion.

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

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

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

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

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

6.7 Computer Assisted Music and Acoustics Research

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

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

Its address is:

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

Teaching

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

Music 253	Introduction to Musical Information	`http://www.stanford.edu/class/music253/`
Music 254	Seminar in Music Representation	`http://www.stanford.edu/class/music254/`

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

Lab for Music Notation and Analysis

CCARH maintains a lab for applications in music notation, analysis, and sound sequencing. It also maintains some CD-ROM titles related to music research.

Currently supported third-party applications include:

Notation software and fonts		
Score	(Windows 9*)	notation and publishing
Finale	(Mac)	notation: MIDI capture and playback
Sibelius	(Windows)	notation: MIDI capture and playback
capella	(Windows)	notation: MIDI capture and playing: pedagogical applications
NoteWriter	(Mac)	notation (20th-century)
FinalScore	(Windows)	conversion from *Finale* to *Score*
St. Meinrad's	(Windows. Mac)	font for Gregorian chant
BachFont	(Windows)	font for musical symbols and symbol groups in text
ChordSymbol	(Windows)	font for analytical markup of scores
SmartScore	(Windows)	optical recognition

Analysis Software		
Humdrum Toolkit	(Linux)	c.70 tools for data query, analysis, visualization, and playback

Sequencer Software		
Cakewalk	(Windows)	MIDI capture, editing, playback, visualization

CD-ROMS
Thesaurus Musicarum Latinarum
CANTUS
Zarlino: *Tutte le opere*
Thesaurus Linguae Graecae
PHI Latin Databases

Research in Data Development, Access, and Query

- **Raw Data**

  Musedata is the host format. Client formats include

  - MIDI (in two flavors–one for sound, one for notation)
  - **kern (for analysis using *Humdrum*)
  - See `http://www.ccarh.org/musedata/`

- **Data Query**

  Themefinder (20.000 incipits)

  - Classical and folk repertories
  - Five levels of musical detail permitted in searches: Boolean searches among levels.
  - See `http://www.themefinder.org/`

Performing Materials

- 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)

Publications

- *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)
- Various occasional publications: http://www.ccarh.org/publications/

6.7.2 The Musical Acoustics Research Library

Gary P. Scavone and Max V. Mathews

The Musical Acoustics Research Library (MARL) is a collection of independent archives or libraries assembled by distinguished groups or individuals in the field of musical acoustics research. MARL is directed by representatives of each member 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.

6.7.3 Web-Based Infrastructure for Research and Teaching

Julius Smith

Web evolution has reached the point where it is now extremely attractive as a basis for educational infrastructure. Advantages of Web-based publications include

- Live demos (Java applets, sound examples).
- Links to related documents anywhere on the Web.
- References are "click and go".
- Printable versions can be offered for download.
- Reachable by Web search engines.
- "Meta-data" available to improve search engine results.

While these advantages are almost as well known as the Web itself, full utilization of them in online publishing is only beginning. It is plausible to imagine that, eventually, Web-based publications will become *primary* in fields such as computer music in which dynamic content is extremely useful. Some implications of Web-based publication are outlined in the online document

"Implications of the Web for Academic Publishing"[5]

Another reason for the slow appearance of Web-based publication may be the time required to prepare documents for the Web. To address this issue, tools for conveniently managing online documents generated from LaTeX are being developed. Current versions are described (and provided) in the online home page

"Tools for Publishing LaTeX Documents on the Web"[6]

Using these tools, 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.

When writing online documents, it is useful to draw upon a collection of *links* to related materials, so that the document will be well connected on the Web. To build such a collection, the beginnings of an online, interlinked "knowledge base" in the field of digital signal processing applied to music and audio is under development. The current state can be seen via the online index

[5] http://www-ccrma.stanford.edu/~jos/webimp/
[6] http://www-ccrma.stanford.edu/~jos/webpub/

"JOS Global Index"[7]

The Global Index was initially generated automatically as a list of links to all HTML pages under the JOS Home Page.[8] As such, it functions as a kind of hypertext *glossary* for web-resident content in signal processing applied to music and audio, particularly for content resident at CCRMA.

A suitable subset of links from the JOS Global Index have been contributed to the Open Dictionary[9], which provides a means for researchers and educators in related fields to organize their respective links into one large "meta-encyclopedia".

In general, the best link targets tend to be "Concept Home Pages" (CHP) — Web destinations devoted 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*[10] 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 integral to CCRMA teaching and research.

[7] http://www-ccrma.stanford.edu/~jos/jospubs.html
[8] http://www-ccrma.stanford.edu/~jos/
[9] http://www.opendict.org
[10] http://www-ccrma.stanford.edu/~jos/resample/

7 Recordings

Recordings of works realized at CCRMA include the following:

- *Computer Music @ CCRMA vol. I.* Works of Aguiar, Norton, Lopez-Lezcano, Hopkins, Hind, and Roy. Produced by Celso Aguiar and Marco Trevisani at CCRMA, Stanford University, 1997.
- *Computer Music @ CCRMA vol. II.* Works of Trevisani, Landa, Soley, Edwards, Dong, and Brummer. Produced by Celso Aguiar and Marco Trevisani at CCRMA, Stanford University, 1997.
- *Center for Computer Research in Music and Acoustics*, CDCM, Computer Music Series, Vol 8. Works of Chafe, Schindler, Jaffe, Berger, and Morrill. Digitally mastered CD, CRC2091, Centaur Records, 1991.
- Matthew Burtner, *Portals of Distortion: Music for Saxophones, Computers, and Stones*. Innova Records (Innova 526), 1999.
- *John Chowning*. – Phoné, Turenas, Stria, and Sabelithe. Digital compact disk, WER2012-50 Wergo, Germany, 1988, Harmonia Mundi, Distributors.
- *Computer Music from CCRMA, vol. I.* Digitally mastered cassette with works by Moorer, Schloss, Schottstaedt, Chafe, Jaffe, Berger, and Chowning. Produced by Jan Mattox for CCRMA, 1982.
- *Computer Music from CCRMA, vol. II.* Digitally mastered cassette with works by various composers, 1984 (out of print).
- *Current Directions in Computer Music - Sound Examples.* Digital compact disk to accompany the book *Current Directions in Computer Music*, J. R. Pierce and M. V. Mathews, editors, MIT Press, Cambridge, MA, 1989.
- *Dinosaur Music.* Digital compact disk. Jaffe – "Silicon Valley Breakdown," Chafe – "In A Word," Schottstaedt – "Water Music I & II," and "Dinosaur Music," WER2016-50, Wergo, Germany, 1988, Harmonia Mundi, Distributors.
- *Intercambio Exchange.* Digital compact disk containing computer music from Buenos Aires and California. Works of Cetta, Chafe, Dobrian, Willey, Losa, Krupowicz, Calzon, Lopez-Lezcano, Pozzati, Jaffe, Labor, Cerana, Linan, Lyon and Viera. Produced by CCRMA, LIPM, and CRCA, 1994.
- *Computer Music Journal Volume CD.* – digital compact disk to accompany the 20th Anniversary Issue includes Chowning – "Turenas", MIT Press, Cambridge, MA, 1996.
- *The Virtuoso in the Computer Age-V.* Music for Mathews/Boie Radio-Drum and Radio Baton, CDCM Vol. 15, CRC 2190 Centaur, 1994. Works by Jaffe ("Terra Non Firma"), Jaffe/Schloss ("Wildlife"), Radunskaya ("A wild and reckless place"), Austin ("Mathews Epsiode"), and Appleton ("Pacific Rimbombo").
- *New Music for Orchestra.* VMM 3024, Vienna Modern Masters, 1994. Works by Jaffe ("Whoop For Your Life!") and others.
- *The Digital Domain.* Elektra/Asylum Records 9 60303-2, 1983. Works by Jaffe (Finale to "Silicon Valley Breakdown"), McNabb ("Love in the Asylum"), Schloss ("Towers of Hanoi"), Mattox ("Shaman"), Rush, Moorer ("Lions are Growing"), and others.
- David Jaffe. *The Seven Wonders of the Ancient World*, – digital compact disk available in September, Well-Tempered Productions, 1996.
- David Jaffe. *XXIst Century Mandolin - acoustic and computer music for the mandolin.* WTP5164, Well Tempered Productions, 1994, Allegro Records, Distributors.

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

- 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.

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

- *II SBCM (II Brazilian Symposium on Computers and Music).* – digital compact disk containing Celso Aguiar – "Piece of Mind", DISC MFG., INC., BHS1046, Brazil, 1996.

- *The Science of Musical Sound - Musical Examples.* Cassette tape produced by Jan Mattox for CCRMA to accompany the book *The Science of Musical Sound*, J. R. Pierce, Scientific American, 1985.

- *Unknown Public (04): Musical Machinery.* – digital compact disk containing Nicky Hind – "Rain". Unknown Public, UPCD04, United Kingdom, 1994.

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