

CENTER FOR COMPUTER RESEARCH IN MUSIC AND ACOUSTICS  
DEPARTMENT OF MUSIC, STANFORD UNIVERSITY  
REPORT NO. STAN-M-107

# CCRMA OVERVIEW

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EDITED BY  
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# 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, including exchange concerts with area computer music centers and an annual outdoor computer music festival in July. In-house technical reports and recordings are available, and public demonstrations of ongoing work at CCRMA are held periodically.

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

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

Support for CCRMA has been received from the late Doreen B. Townsend, Walter Hewlett, the California Arts Council, the Ann and Gordon Getty Foundation, the Mellon Foundation, the National Endowment for the Arts, the National Science Foundation, the Rockefeller Foundation (for artists-in-residence), the System Development Foundation, Apple Computer, ATR Human Information Processing Research Labs, Aureal Semiconductor, Bio Control, Crystal Semiconductor, Digidesign, Dynacord, E-mu, Fast Mathematical Algorithms and Hardware, Fender Musical Instruments Corporation, Hewlett Packard, IBM Computer Music Center, Interval Research, ITRI CCL Taiwan, Kind of Loud Technologies, Korg, Matsushita, Media Vision, NEC, NeXT Computer, Nokia Group, 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.<sup>1</sup> 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)
nando	Fernando Lopez-Lezcano	System Administrator / Lecturer
jay	Jay Kadis	Audio Engineer / Lecturer
bil	William Schottstaedt	Research Associate
gary	Gary Scavone	Technical Director
cnichols	Charles Nichols	Interim Technical Director
vibeke	Vibeke Cleaver	Administrative Assistant
jc	John Chowning	Professor of Music, Emeritus
lcs	Leland Smith	Professor of Music, Emeritus
jrj	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

### 2.2 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
arvinhd	Arvinhd Krishnaswamy	PhD Electrical Engineering
jacobliu	Yi-Wen Liu	PhD Electrical Engineering
vickylu	Hui-Ling Lu	PhD Electrical Engineering
harv23	Harvey Thornburg	PhD Electrical Engineering

<sup>1</sup><http://www-ccrma.stanford.edu/CCRMA/HomePages.html>

## 2.3 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
pdelac	Patricio de la Cuadra	PhD Computer-Based Music Theory and Acoustics
gurevich	Michael Gurevich	PhD Computer-Based Music Theory and Acoustics
pph	Patty Huang	PhD Computer-Based Music Theory and Acoustics
randal	Randal Leistikow	PhD Computer-Based Music Theory and Acoustics
unjung	Unjung Nam	PhD Computer-Based Music Theory and Acoustics
cnichols	Charles Nichols	PhD Computer-Based Music Theory and Acoustics
norton	Jonathan Norton	PhD Computer-Based Music Theory and Acoustics
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
rswilson	Scott Wilson	PhD Computer-Based Music Theory and Acoustics
leigh	Leigh VanHandel	PhD CCRMA-cology
oded	Oded Ben-Tal	DMA Composition
mburtner	Matthew Burtner	DMA Composition
ching	Ching-Wen Chao	DMA Composition
falk	Chris Falk	DMA Composition
cwjones	Christopher Jones	DMA Composition
jugend	Nurit Jugend	DMA Composition
dkeller	Damián Keller	DMA Composition
skoch	Sven-Ingo Koch	DMA Composition
senylee	Seungyon-Seny Lee	DMA Composition
bobby	Bobby Lombardi	DMA Composition
jugend	Jugend Nurit	DMA Composition
cprestia	Chrysa Prestia	DMA Composition
vigorous	Keatly Haldeman	MA Science and Technology
hipple	Aaron Hipple	MA Science and Technology
jdmiller	Joel Miller	MA Science and Technology
mknorris	Molly Norris	MA Science and Technology
perella	Steven Perella	MA Science and Technology

## 2.4 Undergraduate Students

Login	Name	Degree Program
jbyron	Jeffrey Byron	Music, Science and Technology
dkc	David Chisholm	Music, Science and Technology

## 2.5 Visiting Scholars

Login	Name	Home Affiliation	Term
bau	Marcia Bauman	Composer, USA	ongoing
fdc	Joanne Carey	Composer, USA	ongoing
jan	Jan Chomyszyn	Psychoacoustic Researcher, Poland	ongoing
mo	Maureen Chowning	Singer/Composer, USA	ongoing
iwatake	Toru Iwatake	Composer/Researcher, Japan	2000-2001
daj	David Jaffe	Composer/Engineer, USA	ongoing
peer	Peer Landa	Composer, Norway	ongoing
lew	Stephen Lew	Researcher, England	2000-2001
eig	Enrique Moreno	Researcher, Mexico	ongoing
muller	Carl Muller	Researcher, USA	ongoing
pasi	Fiammetta Pasi	Composer, Italy	ongoing
juanig	Juan Reyes	Composer/Researcher, Columbia	ongoing
patte	Patte Wood	ICMA, USA	ongoing
verplank	Bill Verplank	Researcher, USA	2000-2001

## 2.6 Collaborators

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

## 2.7 Industrial Affiliates

Company	Address
Digidesign	Palo Alto, CA
Fender Musical Instruments Corporation	Scottsdale, AZ
Hewlett Packard	Palo Alto, CA
IBM Computer Music Center	Yorktown Heights, NY
Kind of Loud Technologies	Santa Cruz, CA
Nokia Group	Helsinki, Finland
NTT Basic Research Labs	Kanagawa, Japan
Yamaha Corporation	Hamamatsu-shi, Japan





### 3 Facilities

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

The CCRMA computing environment is supported by more than 40 machines that include 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 Waves L2 UltraMaximizer, a Lexicon 224XL digital reverberator, an Eventide Orville processor, Westlake BBSM-10 and JBL 4206 monitors, and outboard gear including equalizers, LA-2A and 1176 compressors, and digital effects processors. A 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 Studio C is organized around a PowerMac G4 computer and an Opcode Studio 5 MIDI interface/MIDI patcher. There is a Yamaha KX-88 weighted-key controller and MIDI equipment including Yamaha SY-99 and VL-1 synthesizers, TX-802 module, Korg Wavestation A/D and X3R modules and Wavedrum synthesizer, E-Mu Proteus/2 module and ESI-32 sampler, and Kurzweil K2000R. There is a Yamaha Disklavier upright piano as well. The Studio C audio system includes a Mackie 24-8 analog mixer, Tascam DA-38, Panasonic SV-3700 DAT recorder, Denon DN-600F CD player, and ProTools MIXplus with 888 I/O and many TDM plug-ins. Monitoring is via four JBL LSR-28P powered speakers. Signal processing is available from a Korg A-1 multi-effects processor. A Plextor 8/20 CD writer is part of the studio as well and CD-Rs can be written from Toast and Jam software from files edited in ProTools or Peak programs.

Studio E is a ProTools-based room with some MIDI capability. Audio equipment includes a Tascam DA-88 w/ IF-88AE AES/EBU converter, Allen and Heath GL2 mixer, and Genelec 1030A monitors. The ProTools system running on a PowerMac G3 features a ProTools—24 core and an 888 I/O module. Several ProTools TDM plug-ins are available and may be shared by Peak software. MIDI equipment includes an E-Mu Emulator IV, Korg X3R, and a Kurzweil K2000 keyboard connected to an Opcode

Studio 5LX interface. A Linux system is available with Sonorus audio card providing 8-channel digital I/O. Monitoring is also available on 4 Ramsa WS-A200 speakers powered by Yamaha P-2200 amps.

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

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

## 4 Courses

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

Prospective graduate students especially interested in the work at CCRMA should apply to the degree program at Stanford most closely aligned with their specific field of study, e.g., Music, Computer Science, Electrical Engineering, Psychology, etc. Graduate degree programs offered in music 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 exercises.

- **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 250. Computer-Human Interaction Technology.**  
A series of lectures, labs and project assignments aimed at introducing enough technology so that students from a mix of disciplines can design and build innovative interface devices. Focuses less on theory and more on practical skills leading to a four-week project: designing and building a working controller.
- **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<sup>2</sup>. 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.

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<sup>2</sup><http://www-ccrma.stanford.edu/>

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

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

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

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

- **Introduction to 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**, **SAM-BOX**, **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

- *fortepiano* (2000) for 4-channel tape utilizing the piano as a sound source.
- *How silent comes the water* (2000) for solo piano  
*How silent comes the water* was commissioned by the Israeli pianist Michal Tal.
- *Saraband* for flute, clarinet, violin, cello, piano, percussion, and computer generated sound (2000)
- *Socratic Dialogues* for violin, percussion, and 8 winds (1999)
- *Soliloquy* for cello and computer generated sound (1999)

### Jonathan Berger

- *My Lai* (2001) for solo piano  
*My Lai* was premiered at the United Nations General Assembly, January 24 2001.
- *Echoes of Light and Time* (2000)  
A sound installation and collaboration with Dale Chihuly. Commissioned by the City of Jerusalem and Chihuly Studio for *Chihuly in the Light of Jerusalem*, 2000.  
Echoes of Light and Time, Berger's millennium collaboration with sculptor Dale Chihuly was heard by over two million visitors and received international attention. The CD adaptation of the work was released by Vitosha/Sony in January 2001.
- *Marcatto Sempre* (2000) for clarinet (commissioned by Ensemble Kaprizma).
- *Miracles and Mud* (1999) for String Quartet.
- *Of Hammered Gold* (1999) for period instruments and digital bird organ (Chamber Music America Millennium Commission).  
Jonathan Berger is a National Endowment for the Arts Fellow, 1996-1997 and recipient of the Chamber Music America Millennium Commission.

### Chris Burns

- *Misprision* (2001) for flute and guitar  
*Misprision* is dedicated to Joanna and Inouk Demers.
- *Gineman* (2000) for harpsichord

In the summer of 2000 I spent almost a month in Bali, studying music and attending performances. When I returned home my head was full of Balinese music, and seemingly nothing else - every shred of music I imagined was in pentatonic scales and cyclic patterns.

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

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

*Gineman* is dedicated to Mary Francis.

- *Fabrication* (2000) for trumpet and electronics

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

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

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

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

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

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

I am indebted to Philip Yampolsky, for his encyclopedic knowledge of Indonesian music, and for his field recordings (available on the Smithsonian Folkways label), which are a source of wonder and delight. Philip's friendship is even more valuable, and *78* is dedicated to him, as one more for his collection.

- *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.

- *Escuela* (1999) for piano and interactive electronics

*Escuela* is the second in a series of piano pieces whose titles refer to places where I've lived - in this case, my first home in California, on Escuela Avenue. The title also reflects the piece's status as one of the first products of my work as a graduate student.

In *Escuela*, a computer is employed to modify the sound of the piano during the performance. The performer controls this process from the piano keyboard, changing the electronic transformations over the course of the piece. The computer applies ring modulation, a classic 1960s technique, to the piano sound, multiplying the number of pitches above and beyond what the pianist plays. These additional, phantom pitches are chosen to reflect the symmetrical pitch structures used in composing the pianist's material. The result is a kind of mirroring - the electronics describe and translate the piano's music in the way that they alter its sound.

Thanks to Juan Pampin, for assistance with the software, and to Christopher Jones, for reading and commenting on early drafts of the piece.

### C. Matthew Burtner

- *Animus/Anima* (2001) for voice and electronics
- *S-Trance-S* (2001) for computer metasaxophone
- *Delta* (2001) for electric saxophone
- *Natigviksuk* (2000) for viola, alto saxophone, piano, and noise generators
- *Studies for Radio Transmitter* (2000) for home-made radio transmitters
- *Oceans of Color* (2000) for 27 solo saxophones
- *Signal Ruins* (2000) for prepared piano, bass drum, and electronics
- *Noisegate 67* (1999/2000) for computer metasaxophone
- *Stone Phase* (1999) for computer-generated tape
- *Frames/Falls* (1999) for amplified violin, amplified double bass, and electronics

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

- *Ping* (2001)

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

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

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

- *Transect* (1999) for CD

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.

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

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

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

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

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

- *Elegy* (2000) for Celletto

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

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

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

- *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-sharp and C (his initials) are highlighted.

- *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.

- *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.

- *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.

#### 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 commissioned 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.

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

- *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.
- *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.

### Seungyon-Seny Lee

- *Je est un Autre II* (2000), for 3 dancers, video images, installations, and computer-generated sounds

*Je est un Autre II* is a journey of the imagination and an aesthetic evolution of its ingredients. The cycle consists of four pieces, each composed for different media. Electro-acoustic music and visual images are combined in the first piece, dance, video images, installations, and computer generated sound in the second, Instrumental sextet and theater in the third, and a sound installation with ceramics in the fourth. In these pieces, Imagination, as a metaphor for the unconscious, signifies the existence which struggles within an ego.

The raw acoustic materials used in *Je est un Autre II* are recordings of the sounds from nature. These sounds were electronically processed, creating a broad palette timbres and sonic textures. These transformed materials are used to develop diverse musical layers, inviting the listener to interpret the music through his own imagination.

The imaginary phase of the unconscious is further represented in the piece by projected images. The images were chosen for their symbolic references to internal psychological states.

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

The installation is intended to convey aspects of Lacan's linguistic structure of discourse. The three panels used as props by the dancers represent the phases of the linguistic process. Discourse originates in the unconscious (represented by the plastic sheet). The abstract form of the idea filters through the memory (the transparent scrim), and is formulated as language (the newspaper sheet).

- *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.

Seungyon-Seny Lee has studied with Jonathan Harvey, and Chris Chafe at Stanford as a D.M.A student. She received a M.M. degree in composition at Boston University where she studied with Lukas Foss and Richard Cornell, and she also has studied with Barry Vercoe at the Media Lab at MIT.

She has composed for both acoustic and electronic means, as well as music for animation and documentary film. In 1999, she collaborated with professor Greg Lam Niemeyer at SUDAC (Stanford Univ. Digital Arts Center) presenting a multimedia installation, "Flythrough," has been performed in Cantor Center for Visual Arts and at the Tech Museum of Innovation in San Jose.

One of her percussion quartet series "Chukwon I" was broadcast by Czech Radio channel Vltava where entered for Musica Nova 99. It has been performed in US, Germany, and Italy at festival such as Internationale Musikinstitut Darmstadt, XIII CIM (Colloquium on Musical Informatics) in l'Aquila. "Chukwon I" was also awarded honorable mention in the Luigi Russolo Competition 2000. Her multimedia series "Je est un Autre III" has been performed in Stanford, California, and at Acanthes, Concours internationaux de la ville de Paris, in Avignon, France.

As a resident artist, she has been in Paris at Cite Internationale des Arts program, and she received a residence award in Djerassi Resident Artists Program in 2001 in Woodside, CA.

In the present, Seungyon-Seny Lee is a composer in a year-long course in composition and computer music at IRCAM (Institut de Recherche et Coordination Acoustique/Musique) in 2000-2001.

### **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 ..." fl.

*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 sound-file 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.

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

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

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

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

The sustained notes are animated with trills and vibratos of three different speeds, flutter tongues, and sung pitches, which create interference with the timbre of the flute.

The timbre of the flute is further processed with computer programming, using amplitude- and ring-modulation, and spatialized around four speakers.

## Juan Reyes

- *Oranged (lima-limon)* for tape

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

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

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

- *ppP* for piano and tape

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

Born in Barranquilla Colombia, Juan Reyes holds degrees in Mathematics and Music Composition. Since 1989 he has co-organized the International Contemporary Music Festival and periodic electro-acoustic cycles in Bogota. He was also professor of art, music and a research associate at La Universidad de Los Andes in Bogota. Currently at CCRMA, Center for Computer Research in Music and Acoustics of Stanford University, his research topics include physical modeling, and spectral modeling of musical instruments, algorithmic composition and their use for expression modeling. Among his works are *Equus* and *Resonancias*, a collection of computer music works for the stage and also several compositions based upon instrumental sound subjects like *Boca de Barra*, for trombone, and *Sygyfrydo* for cello. In the context of sound installations his works *ppP* and *Los Vientos de Los Santos Apostoles* have been presented in museums and galleries of Colombia. His writings have appeared on several international publications and his music has been performed around the world as part of contemporary music radio broadcasts and festivals.

- *Los Vientos de Los Santos Apostoles*

*Los Vientos de Los Santos Apostoles* is a composition for fixed length models of organ pipes tuned in the Bohlen-Pierce Scale as described by Mathews. Models of pipes are based upon flute models by Vesa Valimaki and Matti Karjalainen also algorithmic-ally described by Perry Cook in the Synthesis Toolkit. The Settings for the performance of this piece include an enclosed space or room; red, blue, green and yellow lighting sources coming from the roof and the sides; Four or more CD Players with track shuffle and automatic repeat; a set of three or more pairs of loudspeakers placed strategically to create illusion of space; mixing console and amplifiers. The CD players can be replaced by a PD: pure data patch emulating sound-playing and enhancing interaction with the use of sensors. This article is a description of this composition roughly perceived as a sound installation on which a listener or group of listeners interact or react to the controls, sensors, interface and sound perceived.

The goal of this composition is to create a perceived composition or the illusion of a composition tailored or customized to the listener taste but according to some constraints developed by the composer. The constraints follow the next mainly technical guidelines: about 45 sound-files can be performed by four or more virtual players (CD - Players or PD-Players). They can be performed sequentially or superimposed. One player can perform all sound-files sequentially depending on various determined order types: least to Max, palindromic or random. The duration in this choice can exceed 75 minutes and therefore is not too desired by most visitors. Superimposed sound-files can benefit of harmonies given by the nine degrees of a diatonic scale built upon the Bohlen - Pierce scale. In this sense counterpoint can be perceived while pitch abstraction and line contrast create generative melodic grammars in the mind of the listener. This idea creates the illusion of composing melodies while mixing different sound-files, its textures and intensity. Only one given visitor is in control of the interaction in the installation although a group or ensemble of visitors might also interact. The actions for interacting are: sound-file choice, sound-file start or stop, loudspeaker choice, panning and mixing.

Once inside the room a visitor is confronted with an array of loudspeakers of different shapes and sizes which resemble the idea of a pipe organ. The light sources give the impression of tinted glass which changes its intensity by the sun's position according to the time of day. Candles can also be added if proper ventilation is provided. Several CD players are available and are arranged in such a way that they will persuade the visitor on interacting with its controls. In the software version sensors replace play, stop, volume and shuffle controls and add proximity and position sensors. The output of the sensors is controlled and transmitted to the Pure Data patch by a Basic Stamp II micro-processor and MIDI Boolean values. One CD will interrupt silence and continue as the visitor wish. This can also be combined thereafter by another track or many more. Rhythmic patterns will be subsequently heard and then tone combination, rhythmic manipulation and phase manipulation. The visitor can then choose to stop or continue any track, to change its volume, to repeat. Future improvements will include signal processing like reverb and sample rate conversion to take advantage of more sensors.

Nine pitch classes of the Bohlen-Pierce scale are spread over the range of five octaves going from roughly F2 in the traditional well tempered scale up to D7. Each pitch class is assigned five or seven rhythmic pattern permutations. Each permutation constitutes a sound-file or a track. CD tracks are then randomly ordered to fill a 74 minute compact disc. In the software case (PD) these five or seven sound-files are chosen by the visitor or player either by choice or by probability distributions (weights). Once the user has selected a sound-file, its performance can be aborted with a fade choice or just cut. If the player's performance is satisfactory and the listener content, sound-files can be manipulated by changing their amplitude, speed, speaker assignment, panning, or proximity. Performance can be stopped at any time. Whether the first player is performing or not is not an issue for adding a second player with similar or different options as the first one. Similarly with a third one and, so on. Rhythmic structures are manipulated by rhythmic superposition and phase. Intervals are created by connecting sounds in one track with the others. Total duration in this experience is expected anywhere from 20 seconds to several minutes.

This piece was part of the mixed media and multimedia exposition TELE-vision at the Museum of Modern Art in Bogota Colombia. As many art pieces in the genre in some respect it fulfilled its expectations while there were some unexpected ones. Light for most part was a problem because many users were not able to see CD controls. Random choice of tracks and sound-files work in extended period and not in the short run since many users always chosen track one. The nature of the organ sound was seldom confused with recordings of real instruments. The idea of instant or immediate composition or gratification was not easily perceived. Only curious users wanted to interact and control sound. Many visitors at once also obstructed and overcrowded the space. The concept of non orthodox temperaments is not easily understood in non-concert environments. Finally, in gallery situations it is very hard to have more than two sound installations.

The above issues point out that Sound installation is a genre in the music domain because it can be treated as a traditional composition in the sense that most of its relevant issues are closely related to musical structure. The idea of generative grammars is very dependent on time and

space and of course its components such as reverberation and Doppler pitch are a very strong issue in contemporary composition. Whether listeners perceive this sort of pieces as music or art is another issue left to musicologists and art critics. A composer should not feel interfering in the art domain because as in this case sound is not treated as an object but rather as an entity with pitch and rhythm. Perhaps the difference is the way these musical structures are projected or diffused in contrast with the way a a recital or concert composition is conceived. Modern HI-FI and computer technologies provide means for this sort of compositions with one particular advantage- interaction with a composition is note one to many anymore but it can also be a one to one relationship.

### **Kotoka Suzuki**

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

*Sift* was commissioned by MATA and is dedicated to Carla Kihlstedt

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

- *Yoei* (1999) for computer-generated tape, 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.



## 6 Research Activities

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

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

### 6.1 Computer Music Hardware and Software

#### 6.1.1 *grani*, a granular synthesis instrument for CLM

**Fernando Lopez Lezcano**

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

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

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

#### 6.1.2 A Dynamic Spatial Sound Movement Toolkit

**Fernando Lopez Lezcano**

This brief overview describes a dynamic sound movement toolkit implemented within the context of the CLM software synthesis and signal processing package. Complete details can be found at <http://www-ccrma.stanford.edu/~nando/clm/dlocsigs/>.

*dlocsigs.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 *locsigs* in new or existing CLM instruments (this is a completely rewritten and much improved version of the old *dlocsigs* that I started writing in 1992 while I was working at Keio University in Japan).

The new *dlocsigs* 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 adjacent 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 headphone or speaker listening environments. In all cases doppler shift is also generated as well as amplitude scaling due to distance with user-defined exponents and ratio of direct to reverberated sound.

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

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

### 6.1.3 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 runs with realtime support (audio and MIDI) on SGI (Irix), Linux, and Windows computer platforms. Generic, non-realtime support has been tested under NeXTStep, but should work with any standard C++ compiler.

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, .aif, and .mat output soundfile formats (as well as 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 compatible Windows port (using Direct Sound API) in June 1998. Numerous improvements and extensions have been made since then.

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

## References

- Cook, P.R. and Scavone, G.P. *The Synthesis ToolKit (STK)*, In Proceedings of the 1999 International Computer Music Conference, Beijing, China, 1999.
- Cook, P. R. *Synthesis ToolKit in C++, Version 1.0*, In SIGGRAPH 1996, Course #17 and 18, Creating and Manipulating Sound to Enhance Computer Graphics, May 1996.

### 6.1.4 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/cm.n.tar.gz>, and <pub/Lisp/snd-4.tar.gz>.

### 6.1.5 Common Music

Heinrich Taube

#### What is Common Music?

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

#### History

Common Music began in 1989 as a response to the proliferation of different audio hardware, software and computers that resulted from the introduction of low cost processors. As choices increased it became clear that composers would be well served by a system that defined a portable, powerful and consistent interface to the myriad sound rendering possibilities. Work on Common Music began in 1989 when the author was a guest composer at CCRMA, Stanford University. Most of the system as it exists today was implemented at the Institut fr Musik und Akustik at the Zentrum fr Kunst und Medientechnologie in Karlsruhe, Germany, where the author worked for five years. Common Music continues to evolve

today at the University of Illinois at Urbana-Champaign, where the author is now a professor of music composition. In 1996 Common Music received First Prize in the computer-assisted composition category at the 1er Concours International de Logiciels Musicaux in Bourges, France.

### Implementation

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

### Synthesis Control

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

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

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

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

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

### Contact

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

## 6.2 Physical Modeling

### 6.2.1 Modeling High Frequency Modes of Complex Resonators Using a Waveguide Mesh

Patty Huang, Stefania Serafin, and Julius O. Smith III

This project was motivated by the need for a high-quality model of a violin body, with reasonable computational cost, in the case where the nonlinear interaction between the bow and string prevents the use of commuted waveguide synthesis. In the current model, a biquad filter bank simulates the important low-frequency resonances. For the complex high-frequency resonances, we use a waveguide mesh embedded with absorption filters to tune the decay times and relative amplitudes of the modes. The

goal of the mesh design is to match certain properties of the mesh response to the violin body response in each critical band, such as mode spacing and average bandwidth, to yield an accurate sounding result above some frequency.

The core of this study is to use the waveguide mesh structure not as a physical modeling tool, but as a psychoacoustic modeling tool. The violin body can be generalized to be any complex resonator, and the waveguide mesh is a computational structure which shows promise to simulate the complex and dense high-frequency modes of an instrument's body in a perceptually accurate way.

## 6.2.2 Toward High-Quality Singing Synthesis with Varying Sound Qualities

Hui-Ling Lu

Naturalness of the sound quality is essential for the singing synthesis. Since 95

In this study, we have focused 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.

In addition to providing flexible pitch and volume controls, the desired excitation model is expected to be capable of changing the voice qualities from "laryngealized" (or "pressed"), to "normal", to "breathy" phonation. We propose to use the LF-model (Fant et al., 1985) for the glottal wave shape in conjunction with pitch-synchronous, amplitude-modulated Gaussian noise, which adds an aspiration component to the glottal excitation. By analyzing baritone recordings, we have found a parametric model for controlling vocal textures in synthesized singing.

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## 6.2.3 Scanned Synthesis

Max V. Mathews, Bill Verplank, and Rob Shaw

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

## 6.2.4 Parameter Manipulation for Composing with Physical Models

Juan Reyes

### Abstract

The problem with Physical Models and their appeal to composers is not merely perceptual or aesthetic. Furthermore is not a question of understanding the physics and parameters of the actual instrument. It is a question of achieving musical textures, realistic articulations and extending the qualities of a family of sounds given by a single characteristic timbre in a Physical Model of an instrument. This can be achieved modeling and manipulating expression parameters for musical gesture. This paper describes some composition techniques for rendering a Computer Music piece using Physical Models as the primary source technique for sound synthesis in non real time environments and parameter manipulation by means of envelopes, randomness and chaotic signals for expressiveness.

### Introduction

In this paper, Musical Expression is the synthesis of meaningful musical gestures which are usually categorized on the frequency domain as vibratos, trills, grace notes, appoggiaturas, dynamic changes, etc. Gestures on the time domain are tempo, rubato, rhythmic patterns and durations. The articulation and phrasing of a sequence of notes depends upon combinations of gestures in the time and frequency domains. A performance of a musical event is the relation among the Physical Model of the instrument, a set of rules which describe a sequence of sounds plus their expressive parameters to play the notes. Expressiveness can be applied on a single note basis or to a whole sequence of notes as in musical phrase.

### Synthesis of Expression

In the composition context, real time is not an issue instead creative ideas are seduced by various degrees of freedom and parameters in the Physical Model algorithm. Although a traditional representation of the acoustical instrument is highly desirable, a non conventional behavior gives new fresh possibilities impossible in the real world by morphing the traditional sound of the instrument into different wave shapes or variations [Smith, 1996]. In order to obtain this desired control over the rendering of a composition, all of the expressive parameters need to be computed a priori therefor providing a score file which is later combined with the synthesis of the modeled acoustics of the sound. These parameters can be computed manually or automatic by means of envelopes, functions or algorithms. Several applications of automatic parameter manipulation by mathematical functions in the fields of Randomness, Chaos, Markov Models, Bayesian Nets and Neural Nets provide a high degree of variety for musical expressiveness.

### Expression Modeling

A practical and flexible interface for rendering meaningful musical expression is part of most sound synthesis software packages. In our research, we have used the Common Lisp family of Computer Music Composition programs developed by Bill Schottstaedt and Rick Taube at CCRMA and known as Common Lisp Music (CLM) , Common Music and Common Music Notation. These programs provide a very effective connection between the Physical Model specifications and Algorithmic Composition. In these environments Expression Modeling can be an optional part of an algorithmic composition or it can be a separate process. Physical Models of the flute, piano, plucked string, clarinet and maraca have been used for this purpose.

### Expression and Random or Chaotic Behavior

Given that acoustic sounds have a tendency to behave as random or chaotic systems, it seems natural to manipulate parameters in this way in order to achieve variety in the synthesis of a musical gesture. When randomness and chaos are applied to spectral parameters they produce a characteristic aggregate noise found in woodwinds or bowed instruments [Chafe, 1995]. When applied to a sequence of pitches, they give elements of unpredictability and constant change [Sapp, 2000]. Randomness depends on probability distributions for obtaining a parameter or value. In musical applications we want discrete or integer values to map to a sequence of notes.

A chaotic signal is generated by a mathematical expression that contains some sort of randomness. The periodicity of the values given by the expression depends upon probability distributions and rules which also are dependent on past or present values. The behavior of these functions show symmetry or tendency causing attraction to the value with the highest periodicity. A chaotic behavior is considered as a quantified level of unpredictability [Schroeder, 1991] which needs to be normalized and mapped to values meaningful and within the bounds for synthesis of Physical Models.

### Examples

Musical gestures can be produced on the beat or event scope but they can also be part of melodic phrases, as articulations of combinations of musical notes. Consequently a parameter for expression must be chosen along, keeping in mind that changes developed, are either perceived as mutations in duration or spectra. A mathematical expression or a set of rules or probabilities give values that change always as function of time. Examples of functions which provide a variety of numbers for musical parameter manipulation in Physical Models are:

The random function  $R[n]$  which returns an integer greater than zero and less than or equal to the given value  $n$ . This function is the basis for creating Random Walks or different ways ordering and sorting of a musical event.

The Henon Map: [Weeks, 1999]

$$X[n+1] = (AX[n])^2 + BY[n] \quad y[n+1] = X[n]$$

is an example of a set of rules for generating periodic, quasi periodic and chaotic signal which can be mapped to a set of  $N$  pitches in a sequence of notes. Careful choice of values  $A$  and  $B$  manages the behavior of the signal

The Restrepo Map: [Restrepo, 1997]

$$X[n+1] = X[n] * (X[n]-1) / X[n-1]$$

is a result of Teager's filter and gives a periodic predictable region plus a chaotic region. Values different to 0 and 1 and on the intervals  $X[0]=0.01, x[1]=1.99$  ;  $x[0]=1.99, x[1]=1.99$  can be mapped to rhythmic and drumming patterns in addition to a sequence of pitches.

The Henon Map and the Restrepo map are two examples of recursive functions that can be mapped to midnote values using the following normalizing function suggested by Craig Sapp [Sapp, 2000]:

$$\text{INT}[(x+1)/2 * 127+1/2]$$

### Results

Phrasing and melodic transformations are another class of music gesture which permit use of chaotic expressions. In this, the relationship among a sequence of musical events, past, present and future also determine expressiveness. A group or combination of notes describes not only a melody but how these notes are tied together [1995, Jaffe, Smith]. There can be a function or rule which conducts how these notes are glued together. Thus it can specify durations of musical events as well as space (silence) between each event.

When a chaotic or Random function is applied to the frequency (pitch) of a sound, different shades of noise are obtained. Nevertheless these values can also be applied to filter coefficients providing musical effects. For example, In the case of the Physical Model of the Piano, they can handle detuning factor, stiffness, hammer strike angle, etc. The advantage of using chaos is that the effect seems more natural and reflects the state (energy or entropy) of a real musical instrument.

### Conclusions

It is important to understand that a musical instrument is not a stable or balanced system in reality. Its behavior and responsiveness depends on various factors and consequently a Physical Model should not be framed as a squared box or a fixed parametric system. The degree of musical expression is proportional to the number of degrees of freedom provided by the Physical Model. Nevertheless the complexity of the model is inverse to its degrees of freedom. Therefore, the more flexible the model the harder to understand and handle. More information and parameter descriptions and documentation

in plain vanilla is needed from its designers. In particular interval sets, boundary conditions and valid parameters.

#### Further work

Further research and testing with toggling of different chaotic signals, randomness and parameters needs to be done. The field of parameter estimation and optimization also brings some ideas as to how to control gestures on Physical Models. From the composition standpoint more timbre manipulation will be done trying to extend the flexibility of Computer Models of Musical Instrument and or acoustics.

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### 6.2.5 Acoustic Research and Synthesis Models of Woodwind Instruments

Gary P. Scavone

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

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

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

Current efforts are being directed toward the development of simplified models of conical woodwind instruments and their 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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### 6.2.6 Realistic and Extended Physical Models of Bowed String Instruments

Stefania Serafin

The focus of this research is to obtain a high quality bowed string synthesizer, which is able to reproduce most of the phenomena which appear in real instruments and also can be extended to provide interesting compositional tools. We built a waveguide bowed string physical model which contains all the main physical properties of real instruments i.e. transversal and torsional waves, model for string stiffness, model for the bow-string interaction and body model.

Our current research consists of finding the parameters to drive the model which give expressive sound quality. This is done by estimating the input parameters from recordings on real instruments and using pattern recognition techniques. Our model runs in real time in Max/MSP and STK and was used by Sile O'Modhain in her dissertation on haptic feedback interfaces, by Charles Nichols for his vBow and by Matthew Burtner with the Metasaxophone.

### 6.2.7 Pattern Recognition Approaches to Invert a Bowed String Physical Model

Stefania Serafin

In physical modeling synthesis it is well known how the input parameters that drive the model represent a fundamental component of the resulting sound. Simple models can sound very realistic when driven by parameters that evolve in time in the same way as in real instruments, while elaborated models sound very synthetic if the driving parameters are stationary in time. In this talk I will present an approach to estimate the input parameters of a bowed string physical model. Different applications will be demonstrated.



## 6.2.8 Digital Waveguide Modeling of Acoustic Systems

Julius Smith

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

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

Digital waveguide filters are obtained (conceptually) by sampling the unidirectional traveling waves which occur in a system of ideal, lossless waveguides. Sampling is across time and space. Thus, variables in a DWF structure are equal exactly (at the sampling times and positions, to within numerical precision) to variables propagating in the corresponding physical system. Signal power is defined instantaneously with respect to time and space (just square and sum the wave variables.) This instantaneous handling of 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.

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### 6.2.9 Synthesis of a Neolithic Chinese Flute

Tamara Smyth, Patricio de la Cuadra, and Chris Chafe

As part of an ongoing project to model the 9000 year old bone flutes unearthed at the Jiahu archeological site in china, we have currently implemented a real-time one-dimensional model of a flute, incorporating previously described techniques for air jet and wind instrument turbulence.

The model is implemented as an external object in Pd (pure data) and allows for real-time control of pitch and breath pressure. In order to interact with the flute, we are pitch-tracking the output of a Theremin and mapping the signal to the flute's control parameters. The result has a poignant incongruity: One of the first electronic music instruments controlling a model of one of the oldest musical acoustic instruments yet discovered.

## 6.3 Digital Signal Processing

### 6.3.1 Linear Prediction Analysis of Voice Under the Presence of Sinusoidal Interference

Yi-Wen Liu, Aaron Hipple, and Kyunsuk Pyun

We are interested in tackling the single channel sound source separation problem of voice and non-voice signals. An interesting task would be to separate singing from instrumental accompaniment, pianos or guitars for example. In that case, it is crucial to make estimation of the glottal source of the voice part in the presence of interfering sinusoids.

The focus of our ongoing research is to study the linear prediction analysis of voice and try to come of with new methods to separate voice and non-voice from a single channel mixture.

Particularly, we've worked on an adaptive linear prediction (LP) analysis framework that is based on the LMS algorithm. The adaptive algorithm is causal, and has the potential of following the statistics of the voice more closely. However, the estimation of the LP coefficients is fluctuating around the optimal solution due to the nature of the LMS algorithm.

### 6.3.2 An efficient and fast octave-band filter bank for audio signal processing of a low-power wireless device

Kyungsuk Pyun

Currently there is an integration between computer and portable wireless device as a hardware technology advances rapidly. In this noisy and power hungry environment, a fast and efficient signal processing front-end to compactly represent the audio signal is important.

The need to perform signal processing at an extremely low power, for applications such as in a portable MP3 player or cellular phone motivates the study of FIR filters having only a few taps with small integer coefficients. For example, digital watches do not have floating point operations.

The proposed octave bank filter bank using Gabor like wavelet consists of 12 bandpass filters covering a frequency range from 1Hz to 4KHz. The signal is stored in an one dimensional buffer before the processing, just long enough to be processed by lowpass and highpass filters, which makes both filters non-causal symmetric ones. The index of the buffer is partitioned so that downsampling by two is done automatically.

What is unique in this approach is the organization of the code, namely one loop going through the samples, performing just a few assignment statements per sample. The proposed algorithm runs on the order of  $N$ , which is much faster than the widely used FFT based algorithm. The only calculations needed are addition and shifting. There are no floating point operations in the algorithm. The resolution of the filter is 6 samples/cycle at the maximum sensitivity of each filter.

### 6.3.3 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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### 6.3.4 The Bayesian Approach to Segmentation and Rhythm Tracking

Harvey Thornburg

Segmentation refers to the detection and estimation of abrupt change points. Rhythm tracking refers ultimately to the identification of tempo and meter, but in this context it refers more generally to the

<sup>3</sup><http://www-ccrma.stanford.edu/CCRMA/Courses/420/>

identification of any higher-level "structure" or "pattern" amongst change points. Therefore the name "rhythm tracking", despite being catchy, is somewhat over-optimistic as to the goals of this project. Nevertheless, it serves until a better one can be found.

A naive approach to rhythm tracking is to use the partial output of a segmentation algorithm (say, the list of change \*times\*) to feed a pattern recognition algorithm which learns the structure of this event stream. What "learning" means depends on the how the data is processed. In the online setting, which is easiest to develop, learning means to predict, i.e. to develop posterior distributions over future change times given those already observed.

There are a number of problems with the naive approach. First, it is open loop. We could, and should somehow "close the loop", i.e. use the tracker's output posterior distributions as priors for the segmentation. This requires a Bayesian framework for the segmentation itself, which will be the \*specific\* focus of this talk. There are really two motivations for developing Bayesian techniques:

1. For the specific problem of rhythm tracking, only a fraction of detected change times are likely to convey rhythmic information. As well as note onsets, there could be expressive timbral and/or pitch fluctuations, interfering signals, and so on, which either do not correspond to rhythmic information or do so less reliably than the onset information. Hence, it is desired that the underlying segmentation method learn to recognize specifically the changes corresponding to the desired structure, a phenomenon known as "stream filtering".
2. Independent of rhythm tracking, there is still the concern that we make efficient use of large data samples. When change times exhibit a regular pattern (i.e. anything that differs from Poisson) this means events in a certain location will give information about those far away. Since any practical (i.e. linear time) approximate change detection scheme uses only short windows with local information in the signal about a given change point, it is desired to communicate more global kinds of information, without resorting to large windows.

Second, structures based only on the change times do not seem to cope well with rests, subdivisions, and like phenomena. For instance, it is unlikely that a given list of interarrival times will simply repeat, so immediately we must cope with the problem of variable length cycles. Hence, lists of groups of rhythmic intervals is not a good intermediate representation. We have to be a bit more clever. A proposed solution, easily adapted to general frameworks for change detection, is to propagate information about the "strength" of each change as well as the change time. Formally, we define a change point as the pair  $T_k, E_k$ , where  $T_k$  is the time of a possible change, and  $E_k$  is an indicator that the change actually occurred. Right now the "strength" is defined as the posterior likelihood  $P(E_k - T_k, E_{k-1}, T_{k-1}, E_{k+1}, T_{k+1}, E_{k-2}, T_{k-2}, E_{k+2}, T_{k+2}, \dots)$ . This definition probably needs refinement. This likelihood may be computed using familiar methods (i.e. subspace tests). Intuitively,  $P(E_k - T_k, \dots)$  relates to information about dynamics, and also encodes the indications of "rests" ( $P(E_k - T_k, \dots) \ll 1$ ). In this representation, encoding of temporal information becomes very easy (the "metronome" model), and all the crucial information gets encoded in the patterns of strengths.

I have focused mostly on the Bayesian framework for \*sequential\* change detection when the model after change is unknown. Here the appropriate prior concerns the \*location\* of the next change point, as well as the prior probability the change has occurred before time zero (I find one can just set this to zero, but it's needed to calculate expected cost). Our stopping rule minimizes an expected cost function involving the event of a false alarm, the total number of miss-alarms, and the number of samples observed. Minimizing the latter also minimizes detection delay. Practically, we must also know something apriori concerning the distribution of models after change. There seem to be two promising approaches: the marginalized approach where we average over the entire space of candidate models (we must have a prior distribution for these models), and the multimodel approach where we sample the alternative model space to get some "representative" set. The way I do this using the fewest models possible is to sample at the "poles" of significance shells. I have found multimodel sampling approaches to work better in practice, but a thorough analysis is still being worked out.

It seems the "strength" prior cannot be integrated here, but we can use it in subsequent "backtracking" using the Bayesian subspace test. That part needs more refinement and discussion.

### 6.3.5 Integrated Online-Offline Methods for Audio Segmentation

Harvey Thornburg

Segmentation refers to the identification of temporally homogeneous regions in a recorded signal. To this end, we discuss three frameworks concerning segment boundaries: online (sequential) detection, offline estimation and offline detection. In particular, a method testing for a parameter in a nested sequence of linear subspaces is developed for offline detection. Two integrated approaches are presented: the first interfaces online detectors in forward and backward scan to identify regions where change points may occur, applying offline methods in a refinement stage; the second pursues a local "predictor-corrector" approach. The latter may be adapted for real-time use with lookahead. Both methods run in approximately linear time, yielding detection/estimation performance approaching that of pure offline estimation despite the inherent quadratic cost of the latter. Applications are shown for speech and other highly transient audio signals.

### 6.3.6 Rhythm Tracking and Segmentation with Priors: A Unified Approach

Harvey Thornburg

Audio segmentation is a fundamental problem in model-based analysis of sound. Segmentation means partitioning the time line into intervals over which the model parameters are relatively constant. Within these intervals, model parameters may vary continuously up to some maximum rate.

In segmentation, a fundamental decision must be made as to whether measured parameter variations are classified as continuous or discontinuous. Errors in this decision can result in noticeable artifacts in model-based resynthesis. For example, portamento in the measured pitch of recorded piano tones is clearly unacceptable.

It is thus important that segmentation be as accurate as possible, and that it make use of all appropriate constraints when the context is known.

We have presented a segmentation algorithm which exploits global information about the temporal pattern of parameter changes. For example, in rhythmically predictable music, the history of prior change events influences the probability of change events at future times. In an offline setting (i.e., out of real time), future change events can also be used to estimate the probability of change at a given point in time. Thus, for example, the segmentation can be carried out iteratively, using past and future change-point information from the previous iteration to optimally detect change points on the current iteration. (This is an example of what is sometimes called a batch relaxation method in system identification.)

To use this global information, we develop a Bayesian segmentation method which makes use of prior likelihoods concerning the location and possible presence of change points. To propagate these likelihoods, we develop a rhythm tracker to operate in tandem with the segmentation method. The rhythm tracker uses a combination of extended Kalman filter and hidden Markov model methods to update prior likelihoods for future change point events given those previously identified in segmentation.

There are basically two frameworks for segmentation: online and offline. In the online case, little or no look-ahead is allowed, while in the offline case, all data are available at all times. Even though the problem setting is usually offline, it is worthwhile to develop online methods in the interest of fast approximation.

The complete segmentation software system includes three separate optimal Bayes estimators covering estimation and detection in the offline setting, and detection in the online setting. The Bayesian formulation of each problem can be described as follows:

1. The problem of offline estimation assumes a fixed data window and a known number of changes: one finds the joint maximum a-posteriori (MAP) estimate of the change point locations using location priors.

2. The problem of offline detection begins with a list of possible change point times and decides which of these times corresponds to actual change points. Given prior probabilities of occurrence, our strategy minimizes posterior misclassification error in testing whether the indicated block parameter of segment model parameters is confined to the appropriate linear subspace.
3. The problem of online detection uses only past and present data to detect the next change point. In the Bayesian framework, we minimize posterior expected cost: the cost function weights the number of observations required for detection, the occurrence of a false detection at the stopping time of the test, and the occurrence of past missed detections. The prior distribution of the next change point enters in the form of a hazard rate, i.e., the probability that a change will occur by the next sample given that no change has occurred yet at the present sample. We choose a time-varying hazard rate to minimize the Kullback-Leibler distance between the induced event distribution and the true location prior.

The following properties motivate our integrated solution: If detection delay is short, the stopping time of online detection gives a biased and somewhat unreliable estimate of the true change time. Furthermore, online detection does not use data past the stopping time. Such disadvantages of online detection are balanced by the advantage of speed. Suppose  $N$  is the number of data samples and  $M$  the number of segments. Offline methods use  $O(MN^2)$  operations whereas online detection uses  $O(N)$ . Since our Bayesian framework uses the rhythm tracker to identify global patterns in change points, we may use local windows (small  $N$ ) for applying the offline methods.

Our integrated solution proceeds as follows. We begin by choosing a parameter ( $M$ ) which refers to the maximum number of change points in the window processed by offline methods. We initialize the window boundaries by iterating the online detection ( $M+1$ ) times, selecting the right boundary as the ( $M+1$ )th stopping time. Next, we run offline estimation to find the intervening  $M$  change points, rejecting some by offline detection. Finally, we move the left boundary to the last detected change point, and repeat. This process iterates until the right boundary exceeds the length of the signal. The average running time of our algorithm is  $O(NM^3)$ . The algorithm operates in real time with average lookahead of  $O(M)$  samples. Good performance is obtained even when  $M=1$ .

Finally, we demonstrate the tandem operation of the Bayesian segmentation/rhythm tracker on a variety of musical examples, comparing performance to results on speech where the rhythmic tendencies are much less prevalent.

### 6.3.7 Identification of a Time-Varying Sinusoidal Model

Harvey Thornburg

A robust time-domain method is presented for nonsequential identification of a sinusoidal model with continuously varying parameters. Initially, a dynamic model is obtained in two stages: First, we fit a time-varying AR model where parameter variations are affixed to a basis. Second, we obtain mode dynamics from the AR model from a modified Kamen's recursion including a novel stabilization method for eliminating unwanted mode parameter oscillations. Sufficient conditions are obtained for convergence of the stabilization method. An efficient Sturm-chain algorithm detects when conditions are violated, whereupon the recursion is reinitialized. Finally, the dynamic model gives rise to a state estimation procedure for amplitude/phase tracking, which is robust to model uncertainties.

## 6.4 Controllers and Musical Instruments

### 6.4.1 The Metasaxophone Project

C. Matthew Burtner

The Metasaxophone Project was formed in 1997 in order to explore applications of the extended saxophone. The project simultaneously pursues research in computer music, composition and performance practice. This demonstration will focus on the computer metasaxophone's use of sensor technology and embedded systems to enable real-time expressive control of virtual strings. Musical examples will be drawn from my recent composition, *S-Trance-S* (2001).

For more information please visit: <http://www.metasax.com/>.

### 6.4.2 TouchSound: Haptics in Sound Editing

Lonny Chu

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

The 2 primary goals of TouchSound are:

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

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

### 6.4.3 The Accordiatron: A New Gestural MIDI Controller

Michael Gurevich

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

#### 6.4.4 The vBow: Experiments in the Haptic Feedback and Physical Model Sound Synthesis of a Virtual Violin

Charles Nichols

The vBow is a haptic musical controller human-computer interface, which simulates the force-feedback of a bow on a violin string, while driving a bowed-string physical model. It is both a musical instrument for expressing the gestures of a performer, and a tool for testing friction and vibration models, and the responsiveness of physical model sound synthesis.

The hardware is based on a single servomotor and cable system. The servo is mounted to a housing, which is fastened to a violin-shaped base, for stability. A guide hole for the cable is drilled through the width of the housing, at the height of where a bow contacts a violin string. The cable passes through the guide hole, and wraps around a capstan, which is secured to the shaft of the servo, and spins within a hole drilled through the depth of the housing. At the top of the housing sits a linear bearing, through which a rod passes. At each ends of the rod are a frog and tip, to which the ends of the cable are secured.

As the performer draws the vBow, the rod passes smoothly through the linear bearing, the cable turns the capstan on the shaft of the servo, and a digital encoder reads the rotations of the servo. The output of the encoder and input of the servo are connected to a data acquisition and servo control card installed in a PC. The servo rotations read by the encoder, and decoded by the data acquisition card, are converted into bow direction and velocity by programming, which uses the data to drive a bowed-string physical model. The programming then sends control data through the servo control card, to the servo, which turns the capstan and engages the cable, to simulate friction and vibration on the virtual bow.

#### 6.4.5 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. A new wind controller based on the same concepts is in the final stages of construction. The design of these controllers was inspired by 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 inputting 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, it 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/>.

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

Bill Verplank

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

### 6.5 Audification of Data

#### 6.5.1 Auditory Representation of Complex Data

Jonathan Berger, Michelle Daniels and Oded Ben Tal

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

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

#### 6.5.2 Sound Waves on the Internet from Real-time Echoes

Chris Chafe, Scott Wilson, Randal J. Leistikow, and David Chisholm

New, no compromise, computer applications for audio will be demonstrated using a simplified approach for high quality music and sound streaming over IP networks. Previously existing systems for streaming digital audio involve a number of trade-offs. Because of transmission bandwidth limitations and best effort delivery, audio signal compression of one form or another is typical. Buffering of data, which often delays a signal by seconds, safeguards against delivery uncertainties. Audio is an unforgiving test of networking - one data packet arrives too late and we hear it. Compromises of signal quality and interactivity have been necessary to avoid this basic fact. Along with our new professional audio

applications we have developed SoundWIRE, a utility which affords an intuitive way of evaluating transaction delay and delay constancy. Its final form is an enhanced "ping" that uses actual sound reflection. A musical tone, such as a guitar pluck, can be created by repeatedly reflecting a digital acoustic signal between two hosts. Using the network delay between these reflections to substitute for a guitar string creates a tone whose stability represents perfectly regular service and whose pitch represents transmission latency. The ear's ability to discern minute differences makes this an unforgiving test of network reliability.

## **6.6 Psychoacoustics and Cognitive Psychology**

### **6.6.1 Evaluation of Masking Models**

**Aaron Hipple and David Merrill**

The goal of our research was to evaluate the performance of different psychoacoustic models for simultaneous masking by inserting them into an existing audio coding framework. Several approaches for creating individual maskers and adding maskers were studied. The two models of individual maskers employed were derived from work by Patterson et. al. and Zwicker. The models for additivity of individual maskers considered were maximum masker, intensity addition, and nonlinear addition (i.e. modified power-law as proposed by Lutfi). To evaluate these subtle differences, we created a Tcl/Tk script to allow us to easily run a subjective ITU-R style blind listening test.

### **6.6.2 The State of the Art and Future Directions in Sound Source Separation**

**Aaron Steven Master**

Given a musical recording of an ensemble, for example a rock band with drums, guitar, and vocals, enthusiasts or engineers might want to obtain just the guitar, just the drums, or just the vocals. This goal, in which one obtains the resynthesis of the component sounds of a mixture signal, when initially given only the combined one- or two-channel signal, is called Sound Source Separation. There are several methods for doing this, though they may be generally divided into "data driven" and "model driven." Methods of both these types are often based on a mimicking of the response of the human auditory system. Though the goal and some general approaches are well-defined, current research has achieved impressive results only when highly constrained. Systems recently presented by Ellis, Klapuri and Virtanen, and Kashino and Murase, will be discussed, as will future directions being pursued at CCRMA and elsewhere.

### **6.6.3 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.

## References

- Lakatos, S., Cook, P. R., and Scavone, G. P. *Selective attention to the parameters of a physically informed sonic model*, Acoustic Research Letters Online, Acoustical Society of America, March 2000.

### 6.6.4 Potential Applications of Linguistic Theory to Music Theory

Leigh VanHandel

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

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

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

## 6.7 Machine Recognition in Music

### 6.7.1 Estimation of Sinusoids in Audio Signals Using an Analysis-By-Synthesis Neural Network

Guillermo Garcia

In this paper we present a new method for estimating the frequency, amplitude and phase of sinusoidal components in audio signals. An analysis-by-synthesis system of neural networks is used to extract the sinusoidal parameters from the signal spectrum at each window position of the Short-Term Fourier Transform. The system attempts to find the set of sinusoids that best fits the spectral representation

in a least-squares sense. Overcoming a significant limitation of the traditional approach in the art, preliminary detection of spectral peaks is not necessary and the method works even when spectral peaks are not well resolved in frequency. This allows for shorter analysis windows and therefore better time resolution of the estimated sinusoidal parameters. Results have also shown robust performance in presence of high levels of additive noise, with signal-to-noise ratios as low as 0 dB.

### **6.7.2 Visualizations of Tonality**

**Craig Stuart Sapp**

Two-dimensional pictures of musical key taken over different time-scales in a piece of music will be presented. Each key is assigned a different color related to the circle of fifths. Long-term key measurements occur at the top of the plots, and the analysis window gradually decreases in width to one beat by the bottom of the plots. The resulting pictures graphically display the hierarchical key structure of the music and how the key relates to the underlying chords.

### **6.7.3 Audio Content-Based Retrieval Methods and Automatic Style Classification**

**Unjung Nam**

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

## **6.8 Historical Aspects of Computer Music**

### **6.9**

#### **6.9.1 New Realizations of Electroacoustic Works**

**Chris Burns**

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

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

In realizing the works anew, it was possible to recover some of the variations and alternatives possible in Lucier and Stockhausen's music.

Alvin Lucier's *I Am Sitting In A Room* is largely known through fixed media: tape or compact disc. However, the score explicitly licenses experiment with the basic process, concluding with the suggestion to make versions that can be performed in real time. With this thought in mind, I undertook to make a live realization of Lucier's work, using Miller Puckette's Pd software. The new realization offered a communal listening experience, created a palpable activation of the room and the environment, and produced surprises in the form of inevitable unintended noises, which knitted themselves into the fabric of the music.

While the Lucier realization was underway I also organized the rehearsals for a much more multifaceted and challenging realization: three performances of Karlheinz Stockhausen's *Mikrofonie I*. The new implementation of the electronics uses the Max/MSP environment to realize bandpass filtering (with discrete frequency steps as in the original analog equipment), and volume and panning controls with a number of ergonomic optimizations. There were many aspects of this realization besides the electronics: the ensemble had to select the necessary percussion implements and order the unfixed sections of the score. In many ways, our realization proved to be the chamber version of the work: we used a relatively small tam-tam, and rehearsed and performed in smaller spaces (including an art gallery and a storefront). With these practical constraints in mind, we traded drama for detail, preferring subtle textures to bold theatrical gestures.

Despite the considerable distance between the new realizations of these two works and the performing traditions established by their composers, there is little possibility of confusing realization with composition. This is precisely the interest of making realizations - the process is an opportunity to engage with another composer's thought. However, performing traditions are an informative context, and not a final arbiter.

In this regard, there are parallels with the recent trend towards the historically informed performance of early music, and particularly recent scholarship regarding the limits of "authenticity." In the realizations under study, the eventual relationship of the new versions to the existing traditions of performance arose as a series of small decisions and practical solutions - as is the case with most musical performances.

## 6.10 Computer Assisted Music and Acoustics Research

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

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

Its address is:

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

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

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

Sequencer Software		
<i>Cakewalk</i>	(Windows)	MIDI capture, editing, playback, visualization

CD-ROMS
<i>Thesaurus Musicarum Latinarum</i>
<i>CANTUS</i>
Zarlino: <i>Tutte le opere</i>
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 repertoires
- 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](mailto: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/>

### 6.10.2 The Musical Acoustics Research Library

Gary P. Scavone and Max V. Mathews

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

#### Background and History

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

In conjunction with the establishment of the Catgut Acoustical Society Library, CCRMA and CAS encouraged the idea of having the personal archives of Arthur Benade and John Backus at the same site. Both were world leaders in the study of wind instrument acoustics. The personal files of John Backus were acquired in 1995. An agreement for the establishment of the Arthur H. Benade Archive at CCRMA was reached in 1997. In autumn 1996, representatives of CAS and CCRMA, together with Virginia Benade, established the Musical Acoustics Research Library at CCRMA in order to provide a single point of reference for the various collections. The most recent addition to MARL, the John W. Coltman Archive, was founded in April 1998. The archives of three of the most prominent wind instrument acousticians of our time, together with the extensive string instrument resources of the Catgut Acoustical Society Library, position MARL as a primary musical acoustics reference source in the world.

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

## Activities

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

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

### 6.10.3 Web-Based Infrastructure for Research and Teaching

#### Julius Smith

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”<sup>4</sup>

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 L<sup>A</sup>T<sub>E</sub>X are being developed. Current versions are described (and provided) in the online home page

“Tools for Publishing LaTeX Documents on the Web”<sup>5</sup>

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.

<sup>4</sup><http://www-ccrma.stanford.edu/~jos/webimp/>

<sup>5</sup><http://www-ccrma.stanford.edu/~jos/webpub/>



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

“JOS Global Index”<sup>6</sup>

The Global Index was initially generated automatically as a list of links to all HTML pages under the JOS Home Page.<sup>7</sup> 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<sup>8</sup>, 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*<sup>9</sup> 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.

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<sup>6</sup><http://www-ccrma.stanford.edu/~jos/jospubs.html>

<sup>7</sup><http://www-ccrma.stanford.edu/~jos/>

<sup>8</sup><http://www.opendict.org>

<sup>9</sup><http://www-ccrma.stanford.edu/~jos/resample/>

## 7 Recordings

Recordings of works realized at CCRMA include the following:

- *Arctic Contrasts*, Min Ensemble. Matthew Burtner, "Sikuigvik". Euridice Records (EUCD#012), 2000.
- *II SBCM (II Brazilian Symposium on Computers and Music)*, – digital compact disk containing Celso Aguiar – "Piece of Mind", DISC MFG., INC., BHS1046, Brazil, 1996.
- Matthew Burtner. *Portals of Distortion: Music for Saxophones, Computers, and Stones*. Innova Records (Innova 526), 1999.
- *Cache*, Compilation of winners of the Young and Emerging Sound Artists competition sponsored by the CEC and SOCAN. Michael Gurevich, "Soft White". Canadian Electroacoustic Community, 2000.
- *Center for Computer Research in Music and Acoustics*, CDCM, Computer Music Series, Vol 8. Works of Chafe, Schindler, Jaffe, Berger, and Morrill. Digitally mastered CD, CRC2091, Centaur Records, 1991.
- Chris Chafe. *Arco Logic*, Solo composer album. Centaur Records, Inc. (CRC2514), 2001.
- Chris Chafe. *Arcology*, Performer/Composer album. Centaur Records, Inc. (CRC2515), 2001.
- *John Chowning*, – Phoné, Turenas, Stria, and Sabelithe. Digital compact disk, WER2012-50 Wergo, Germany, 1988, Harmonia Mundi, Distributors.
- *Current Directions in Computer Music - Sound Examples*. Digital compact disk to accompany the book *Current Directions in Computer Music*, J. R. Pierce and M. V. Mathews, editors, MIT Press, Cambridge, MA, 1989.
- *Computer Music @ CCRMA vol. I*. Works of Aguiar, Norton, Lopez-Lezcano, Hopkins, Hind, and Roy. Produced by Celso Aguiar and Marco Trevisani at CCRMA, Stanford University, 1997.
- *Computer Music @ CCRMA vol. II*. Works of Trevisani, Landa, Soley, Edwards, Dong, and Brummer. Produced by Celso Aguiar and Marco Trevisani at CCRMA, Stanford University, 1997.
- *Computer Music from CCRMA, vol. I*. Digitally mastered cassette with works by Moorer, Schloss, Schottstaedt, Chafe, Jaffe, Berger, and Chowning. Produced by Jan Mattox for CCRMA, 1982.
- *Computer Music from CCRMA, vol. II*. Digitally mastered cassette with works by various composers, 1984 (out of print).
- *Computer Music Journal Volume CD*, – digital compact disk to accompany the 20th Anniversary Issue includes Chowning – "Turenas", MIT Press, Cambridge, MA, 1996.
- *The Digital Domain*. Elektra/Asylum Records 9 60303-2, 1983. Works by Jaffe (Finale to "Silicon Valley Breakdown"), McNabb ("Love in the Asylum"), Schloss ("Towers of Hanoi"), Mattox ("Shaman"), Rush, Moorer ("Lions are Growing"), and others.
- *Dinosaur Music*. Digital compact disk. Jaffe – "Silicon Valley Breakdown," Chafe – "In A Word," Schottstaedt – "Water Music I & II," and "Dinosaur Music," WER2016-50, Wergo, Germany, 1988, Harmonia Mundi, Distributors.
- Janet Dunbar. *Spirit Journey* Original electroacoustic and computer music with spoken and sung poetry and guitar and synthesizer improvisation. Amberlight Productions (JD10SJ), 1999.
- *Electroacoustic Music II*. Music by Berger, Child, Dashow, Duesenberry, Shapiro (Jonathan Berger: "An Island of Tears"), Neuma 450-73-[J].

- *Intercambio Exchange*. Digital compact disk containing computer music from Buenos Aires and California. Works of Cetta, Chafe, Dobrian, Willey, Losa, Krupowicz, Calzon, Lopez-Lezcano, Pozzati, Jaffe, Labor, Cerana, Linan, Lyon and Viera. Produced by CCRMA, LIPM, and CRCA, 1994.
- David Jaffe. *The Seven Wonders of the Ancient World*, – digital compact disk available in September, Well-Tempered Productions, 1996.
- David Jaffe. *XXIst Century Mandolin - acoustic and computer music for the mandolin*. WTP5164, Well Tempered Productions, 1994, Allegro Records, Distributors.
- Michael McNabb. *Computer Music*. Digitally mastered LP. McNabb – “Dreamsong,” “Love in the Asylum,” “Orbital View,” (LP out of print) - CD now available as WER-2020-2, Wergo, Germany, 1994, Harmonia Mundi, Distributors.
- Michael McNabb. *Invisible Cities*. Digital compact disk, WER-2015-50, Wergo, Germany, 1988, Harmonia Mundi, Distributors.
- *Musica Maximalista - Maximal Music Vol. 2*, – digital compact disk containing Celso Aguiar – “Piece of Mind”, CD MM-002, Studio PANorama, Brazil, 1996.
- *Musica Maximalista - Maximal Music Vol. 3*, – digital compact disk containing Celso Aguiar – “All blue, I write with a blue pencil, on a blue sky”, CD MM-003, Studio PANorama, Brazil, 1997. CD of the II International Electroacoustic Music Competition of So Paulo.
- *Musica Nova 2000 Final Protocol CD*. Matthew Burtner, “Fern”, First Prize, Musica Nova 2000 Electroacoustic Music Competition. Society for Electroacoustic Music of Czech Republic, Ministry of Culture, Czech Music Council, 2000.
- *New Music for Orchestra*. VMM 3024, Vienna Modern Masters, 1994. Works by Jaffe (“Whoop For Your Life!”) and others.
- *Night Chains*. (Composers Recordings Inc - Emergency Music Series, CRI CD681). Jeffrey Krieger, electronic cello (Jonathan Berger: “The Lead Plates of the Rom Press”).
- *The Science of Musical Sound - Musical Examples*. Cassette tape produced by Jan Mattox for CCRMA to accompany the book *The Science of Musical Sound*, J. R. Pierce, Scientific American, 1985.
- *Unknown Public (04): Musical Machinery*, – digital compact disk containing Nicky Hind – “Rain”, Unknown Public, UPCD04, United Kingdom, 1994.
- *The Virtuoso in the Computer Age-V*. Music for Mathews/Boie Radio-Drum and Radio Baton, CDCM Vol. 15. CRC 2190 Centaur, 1994. Works by Jaffe (“Terra Non Firma”), Jaffe/Schloss (“Wildlife”), Radunskaya (“A wild and reckless place”), Austin (“Mathews Epsiode”), and Appleton (“Pacific Rimbombo”).

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

## 8 Publications

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

- Berger, J. (1996). Modeling the degree of realized expectation in functional tonal music: A study of perceptual and cognitive modeling using neural networks. In *ICMC (1996)*.
- Berger, J. (1998). Connected to what? *Journal of the College Music Society*, (1). Plenary address, 1997.
- Berger, J. (1999a). The ghost of johannes brahms. In *Proceedings of the TED IX (CD-ROM disk 1)*, Monterey, CA.
- Berger, J. (1999b). Modeling composition, performance and listening. In *Music at the Crossroads: Technology and Composition Pedagogy*, Rubin Academy of Music, Jerusalem. Plenary address. Proceedings published in *Art in Time*, Vol. 4, Issue 1, pg. 53, 1999.
- Berger, J. (2001a). *Musical Expectations*. Cambridge, MA: MIT Press.
- Berger, J. (2001b). *Virtual Music: Computer Synthesis of Musical Style*, chap. Who Cares If It Listens? Cambridge, MA: MIT Press.
- Berger, J., Coifman, R., and Goldberg, M. (1995). Denoising a recording of caruso using local trigonometric bases with lp entropy as cost function. In *Proceedings of the Central Section Meeting of the American Mathematical Society, Chicago, Illinois*. American Mathematical Society. Presentation in special session *Extensions and applications of harmonic analysis: Spaces of homogeneous type and wavelet analysis*.
- Berger, J., Coifman, R., and Goldberg, M. (1996). Using adaptive basis selection to separate noise from music. In *Proceedings of the 16th Workshop on Pure Mathematics: Nonlinear Partial Differential Equations and their Applications*, pp. 133–143, Chuncheon, S. Korea. Pure Math. Research Assoc. and the Korean Academic Council.
- Berger, J. and Cook, P. R. (1999). Sonification report: Status of the field and research agenda. Tech. rep. Prepared for the National Science Foundation by members of the International Community for Auditory Display (ed. G. Kramer *et al.*).
- Berger, J. and Gang, D. (1997). A neural network model of metric perception and cognition in the audition of functional tonal music. In *ICMC (1997)*, p. 158. (Also contained in STAN-M-101).
- Berger, J. and Gang, D. (1998). A computational model of meter cognition during the audition of functional tonal music: Modeling *a-priori* bias in meter cognition. In *ICMC (1998)*, p. 253.
- Berger, J. and Gang, D. (1999). A unified neurosymbolic model of the mutual influence of memory, context and prediction of time ordered sequential events during the audition of tonal music. Tech. rep., Am. Assoc. of Artificial Intelligence Technical Report SS-99-05.
- Berger, J. and Gang, D. (2000). A real time model of the formulation and realization of musical expectations. *Music Perception*. Submitted for publication.
- Berners, D. (1999). *Acoustics and Signal Processing Techniques for Physical Modeling of Brass Instruments*. Ph.D. thesis, Dept. of Electrical Engineering, Stanford University. Available as Stanford University Department of Music Technical Report STAN-M-105 (\$14.00).
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- Chafe, C. (1997). Statistical pattern recognition for prediction of solo piano performance. In ICMC (1997). (Also contained in STAN-M-101).
- Chafe, C. (1999). Interplay(er) machines. *Contemporary Music Review*, 18(3):87-95.
- Chafe, C. (2000). chap. Round Table. Paris: IRCAM. Electronic book.
- Chafe, C. and O'Modhrain, S. (1996). Musical muscle memory and the haptic display of performance nuance. In ICMC (1996). (Also contained in STAN-M-99).
- Cook, P. R. (1996). Singing voice synthesis: History, current work, and future directions. *Computer Music Journal*, 20(3).
- Cook, P. R., editor (1999). *Music, Cognition and Computerized Sound: An Introduction to Psychoacoustics*. Cambridge, MA: MIT Press.
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- Gang, D. and Berger, J. (1998). A computational model of meter cognition during the audition of functional tonal music: Modeling a-priori bias in meter cognition. In ICMC (1998).
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