CENTER FOR COMPUTER RESEARCH IN MUSIC AND ACOUSTICS
DEPARTMENT OF MUSIC, STANFORD UNIVERSITY
REPORT NO. STAN-M-105

# CCRMA OVERVIEW

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EDITED BY
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# Contents

# 1 General Information


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

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

The CCRMA community consists of administrative and technical staff, faculty, research associates, graduate research assistants, graduate and undergraduate students, visiting scholars, visiting researchers and composers, and industrial associates. Departments actively represented at CCRMA include Music, Electrical Engineering, Mechanical Engineering, and Psychology.

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

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

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


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

## 2 Roster

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

### 2.1 Staff and Faculty

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

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

## 2.2 Music Graduate Students

| Login | Name | Degree Program |
|---|---|---|
| cburns | Christopher Burns | PhD Computer-Based Music Theory and Acoustics |
| lonny | Lonny Chu | PhD Computer-Based Music Theory and Acoustics |
| pchordia | Parag Chordia | PhD Computer-Based Music Theory and Acoustics |
| tkunze | Tobias Kunze | PhD Computer-Based Music Theory and Acoustics |
| randal | Randal Leistikow | PhD Computer-Based Music Theory and Acoustics |
| unjung | Unjung Nam | PhD Computer-Based Music Theory and Acoustics |
| cnichols | Charles Nichols | PhD Computer-Based Music Theory and Acoustics |
| norton | Jonathan Norton | PhD Computer-Based Music Theory and Acoustics |
| sile | Sile O'Modhrain | PhD Computer-Based Music Theory and Acoustics |
| juan | Juan Pampin | PhD Computer-Based Music Theory and Acoustics |
| craig | Craig Sapp | PhD Computer-Based Music Theory and Acoustics |
| tamara | Tamara Smyth | PhD Computer-Based Music Theory and Acoustics |
| savd | Scott Van Duyne | PhD Computer-Based Music Theory and Acoustics (on leave) |
| leigh | Leigh VanHandel | PhD CCRMA-cology |
| aguiar | Celso Aguiar | DMA Composition |
| oded | Oded Ben-Tal | DMA Composition |
| mburtner | Matthew Burtner | DMA Composition |
| ching | Ching-Wen Chao | DMA Composition |
| jmd | Janet Dunbar | DMA Composition |
| falk | Chris Falk | DMA Composition |
| rjfleck | Robert Fleck | DMA Composition |
| cwjones | Christopher Jones | DMA Composition |
| junkim | Jun Kim | DMA Composition |
| senylee | Seungyon-Seny Lee | DMA Composition |
| bobby | Bobby Lombardi | DMA Composition |
| jmcginn | John McGinn | DMA Composition |
| kotoka | Kotoka Suzuki | DMA Composition |
| kgiesing | Kristopher Giesing | MA Science and Technology |
| mokun | Michael Okun | MA Science and Technology |
| issac | Issac Roth | MA Science and Technology (on leave) |
| sturm | Bob L. Sturm | MA Science and Technology |
| rachel | Rachel Wilkinson | MA Science and Technology |

## 2.3 Engineering Graduate Students

| Login | Name | Degree Program |
|---|---|---|
| bilbao | Stephan Bilbao | PhD Electrical Engineering |
| dpberner | David P. Berners | PhD Electrical Engineering |
| cedmonds | Christopher Edmonds | PhD Electrical Engineering |
| guille | Guillermo Garcia | PhD Electrical Engineering |
| scottl | Scott Levine | PhD Electrical Engineering |
| vickylu | Hui-Ling Lu | PhD Electrical Engineering |
| purswel | Elizabeth Purswell | PhD Electrical Engineering |
| putnam | William Putnam | PhD Electrical Engineering (on leave) |
| stilti | Timothy Stilson | PhD Electrical Engineering (on leave) |
| harv23 | Harvey Thornburg | PhD Electrical Engineering |
| traube | Caroline Traube | PhD Electrical Engineering |
| yoonie | Yoon Kim | PhD Electrical Engineering |

## 2.4 Undergraduate Students

| Login | Name | Degree Program |
|---|---|---|
| n/a | Vincent Duron | Music, Science and Technology |
| daniel | Daniel Gould | Music, Science and Technology |
| pph | Patty Huang | Music, Science and Technology |
| kkumar | Kevin Kumar | Music, Science and Technology |
| blackrse | John A. Maurer, IV | Music, Science and Technology |
| ajm | Adam Miller | Music, Science and Technology |
| gramps | John Niekrasz | Music, Science and Technology |
| kris | Christine Ohm | Music, Science and Technology |
| jab2and | Jabari Anderson | Music, Science and Technology (minor) |
| jbrooks | James Brooks | Music, Science and Technology (minor) |
| keach | Keach Hagey | Music, Science and Technology (minor) |
| n/a | Chad Hollingsworth | Music, Science and Technology (minor) |
| kakodkar | Siddharth Kakodkar | Music, Science and Technology (minor) |
| n/a | James Kim | Music, Science and Technology (minor) |
| n/a | Matthew Louie | Music, Science and Technology (minor) |
| ericm | Frederick Miller | Music, Science and Technology (minor) |
| entropy | Gabriel Serrano | Music, Science and Technology (minor) |

## 2.5 Visiting Scholars

| Login | Name | Home Affiliation | Term |
|---|---|---|---|
| bau | Marcia Bauman | Lecturer, Music Department | ongoing |
| jdc | Joanne Carey | Composer, USA | ongoing |
| jan | Jan Chomyszyn | Psychoacoustic Researcher, Poland | ongoing |
| mo | Maureen Chowning | Singer/Composer, USA | ongoing |
| diliscia | Oscar Pablo Di Liscia | Music Professor, Argentina | 1/99 – 4/99 |
| fujishim | Takuya Fujishima | Engineer from Yamaha, Japan | 6/98 – 6/99 |
| dang | Dan Gang | Research, Israel | 10/98 – 12/99 |
| n/a | Fabien Gouyon | Researcher, France | 3/99 – 9/99 |
| daj | David Jaffe | Composer, Software Research Engineer, USA | ongoing |
| skjeng | Shyh-Kang Jeng | EE Professor, Taiwan | 1/99 – 7/99 |
| peer | Peer Landa | Composer, Norway | ongoing |
| levitin | Daniel Levitin | Psychology Research/Lecturer, USA | ongoing |
| eig | Enrique Moreno | Researcher, Mexico | ongoing |
| muller | Carl Muller | Researcher, USA | ongoing |
| jcpark | Jong-Chul Park | Prof. of Composition, Korea | 9/98 – 8/99 |
| pasi | Fiammetta Pasi | Composer, Italy | ongoing |
| n/a | Hendrick Purwins | Researcher, Germany | 3/99 – 7/99 |
| serafin | Stefania Serafin | Researcher, Italy | 3/99 – 7/99 |
| shehadi | Charles Shehadi | Media, Art, Music, USA | 10/98 – 8/99 |
| shimoda | Toshiaki Shimoda | Engineer from Kobe Steel, Japan | 8/98 – 7/99 |
| patte | Patte Wood | ICMA, USA | ongoing |
| marek | Marek Zoffaj | Composer, Slovakia | 8/97 – 8/99 |

## 2.6 Collaborators

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

## 2.7 Industrial Affiliates

| Company | Address |
| --- | --- |
| Digidesign | Palo Alto, CA |
| Digital Theater Systems | Westlake Village, CA |
| E-mu Systems | Scotts Valley, CA |
| Hewlett Packard | Palo Alto, CA |
| Interval Research Corporation | Palo Alto, CA |
| Kobe Steel | Osaka, Japan |
| NTT Basic Research Labs | Kanagawa, Japan |
| Opcode Systems | Palo Alto, CA |
| Texas Instruments | Dallas, Texas |
| Yamaha Corporation | Hamamatsu-shi, Japan |

## 3 Facilities

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

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

The MIDI-based systems include various Macintosh computers with Yamaha, Roland and Korg equipment including Yamaha DX, SY, TG and VL synthesizers, KX88 keyboard controller, Korg X3R, I3, WaveDrum and WaveStations, Kurzweil K2000R, E-Mu ESI-32 sampler, and digital delays and reverberation. The mixing console is a Mackie 24-8 in the MIDI Studio and an Allen and Heath GL2 in Studio E, while multitrack recording is supported by DA-38 digital 8-track recorders. Speakers are Ramsa WS-A200: stereo with subwoofer in MIDI studio and quad in Studio E. Studio E is also equipped with Genelec 1030A monitors. Other equipment available includes IVL pitch trackers, a Buchla Lightning MIDI controller, several Mathews Radio Drum controllers, two Opcode Studio 5 MIDI patchers, and drum machines from Yamaha.

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

The digital editing studio (Studio D) consists of a SGI 02 with 8-channel digital I/O, Dyaxis II digital editing system, Digidesign Sound Designer II system, Meyer 833 monitor speakers, a Yamaha DMP-7D digital mixer, Yamaha SPX-1000 processor, a Panasonic SV-3700 DAT recorder and Macintosh and NeXT computers. Digital and analog patchbays provide interconnection.

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

Emulator IV.

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. A graphical environment for real-time DSP research, SynthBuilder, is also used under NEXTSTEP. Of course there is a wide variety of public domain software for text, image and sound processing installed on all workstations.

# 4 Courses

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

Prospective graduate students especially interested in the work at CCRMA should apply to the degree program at Stanford most closely aligned with their specific field of study, e.g., Music, Computer Science, Electrical Engineering, Psychology, etc. Graduate degree programs offered in music are the MA in Music, Science, and Technology; the DMA in Composition; and the PhD in Computer-Based Music Theory and Acoustics. Acceptance in music theory or composition is largely based upon musical criteria, not knowledge of computing. Admission requirements for degree programs can be obtained directly from each particular department. CCRMA does not itself offer a degree.

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

## 4.1 University Courses at CCRMA

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

Courses offered at CCRMA include:

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

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

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

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

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

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

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

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

- **Music 152. Musical Acoustics.**

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

- **Music 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.

  - **192B. Advanced Sound Recording Technology.**

    Topics: digital audio including current media, formats, editing software, and post-processing techniques. Also, microphone selection and placement, grounding and shielding techniques, noise reduction systems and advanced multi-track techniques.

  - **192C. Session Recording.**

    Independent engineering of recording sessions.

- **Music 220. Computer-Generated Music**

  - **Music 220A. Fundamentals of Computer-Generated Sound.**

    Techniques for digital sound synthesis, effects, and reverberation. Topics: summary of digital synthesis techniques (additive, subtractive, nonlinear, modulation, wavetable, granular, spectral-modeling, and physical-modeling); digital effects algorithms (phasing, flanging, chorus, pitch-shifting, and vocoding); and techniques for digital reverberation.

  - **Music 220B. Compositional Algorithms, Psychoacoustics, and Spatial Processing.**

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

  - **220C. Seminar in Computer Music Research.**

    Individual projects in composition, psychoacoustics, or signal processing.

  - **220D. Research.**

    Independent research projects in composition, psychoacoustics, or signal processing.

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

  Various topics according to interest.

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

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

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

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

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

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

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

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

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

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

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

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

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

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

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

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

## 4.2 Workshops

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

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

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

  This course covers analysis and synthesis of musical signals based on spectral and physical models. It is organized into morning lectures covering theoretical aspects of the models, and afternoon labs. The morning lectures present topics such as Fourier theory, spectrum analysis, the phase vocoder, digital waveguides, digital filter theory, pitch detection, linear predictive coding (LPC), and various other aspects of signal processing of interest in musical applications. The afternoon labs are hands-on sessions using SMS, the Synthesis Toolkit in C++, SynthBuilder, and other software systems and utilities. The lectures and labs are geared to a musical audience with basic experience in math and science. Most of the programs used in the workshop are available to take.

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

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

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

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

- **Interactive Composition and Performance with Computers**

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

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

- **Advanced Projects in Algorithmic Composition**

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

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

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

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

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

- **Intensive Audio Digital Signal Processing**

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

# 5 Compositional Activities

## 5.1 Overview

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

Continuity has been maintained over the entire era. For example, scores created on the PDP10 or Samson Box have been recomputed in the NEXTSTEP computing environment, taking advantage of its increased audio precision. To summarize all these names for CCRMA's composing environment, the synthesis instrument languages have been, in chronological order, **MUS10, SAMBOX, CLM/MusicKit** and the composing language succession has been **SCORE, PLA, Common Music/Stella**. Other computers and software are also used for composition. Several composers have realized pieces which make extensive use of MIDI equipment, especially Yamaha synthesizers controlled via Macintosh computers. The acquisition of a Dyaxis II Digital Audio Processor and several Macintosh II computers has brought renewed interest in real-time control and computer-based "musique concrète." The programming environments being used for composition and developmental research for all these systems include **MAX, 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 Greece, Hong Kong, 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:

**Celso Aguiar**

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

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

- *Monologue for Two*

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

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

**Oded Ben-Tal**

- *Medea* (1998) for three singers and chamber ensemble.

  This is a revised and expanded version of a piece I started writing in 1996. The first version was performed in Jerusalem and the current version is scheduled for concert at Stanford, Feb. 15 1999.

- *In memory of Federico Garcia-Lorca* (1998) - four short pieces for percussion solo.

  Oded Ben-Tal is currently working on a piece for violin, percussion and 8 wind instruments, as well as a short cello and tape piece.

### Jonathan Berger

- *Miracles and Mud* (1999) for String Quartet.
- *Of Hammered Gold* (1999) for computer, harpsichord, Baroque flute, Baroque oboes, Baroque violin and gamba (Chamber Music America Millenium Commission).
- *Arroyo* (1998) for motion detected dancer, computer and instruments.
- *Elegy* (1998) for alto and chamber ensemble.
- *Con Carne* (1998) for Disklavier and pianist.

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

### Chris Burns

- *Calyx* (1998)

  *Calyx* is the most recent in a series of pieces exploring the possibilities of a set of filters connected in an infinite feedback loop. No initial stimulus to the filters is necessary – the internal noise of the system, amplified endlessly via the loop, is sufficient to produce a rich palette of sounds.

  The sonic results of this process are entirely context-dependent; the musical possibilities at each moment are limited by the state of the loop at the previous moment. As the filter settings are changed in real-time, the sounds produced by the loop can be shaped into musical forms. *Calyx* is a recording of such a real-time "performance," controlled by computer and refined over a period of months.

- *Escuela* (1999)

  *Escuela* is the second in a series of piano pieces which somehow refer to places where I've lived – in this case, my first home in California, on Escuela Avenue. In this case, the piece is (almost inevitably) bound up in my early experiences as a graduate student, thereby enriching the double meaning of the title.

  In *Escuela*, a computer is employed to modify the sound of the piano in real time. The performer controls the software from the piano keyboard, applying ring modulations which precisely reflect the pitch structure of the original piano music. The result is a kind of mirroring – at a microscopic level, the electronics describe the piano's music in the way that they alter its sound.

### C. Matthew Burtner

- *Kuniklu̇k* for ensemble and noise generator (1999)
- *Rhythm/Noise Study (in Metal)* for computer-generated tape (1998)
- *Frames/Falls* for amplified violin, amplified double bass, and computer-generated tape (1998)
- *Portals of Distortion* for nine tenor saxophones (1998)

  A solo CD of new music by Matthew Burtner, "Portals of Distortion: Music for Saxophones, Computers, and Stones" was released in January, 1999 by Innova Records (Innova 526).

### Joanne D. Carey

- *Adventures on a Theme* (1997)

    1. The Unraveling
    2. Topsy Turvey and Haywire
    3. Harmony Rains

    *Adventures on a Theme* for flute and radio-baton is a flute concerto whose synthesized orchestra includes singing voices, marimba, percussion and guitar as well as strings, woodwinds and brass, often in unusual combinations. Although there is no story-line associated with it, the music seems to tell a story in its winding and wayward lyricism. The middle movement, "Topsy Turvy and Haywire", which is entirely improvised, presents another view of the protagonist, the theme. From a compositional standpoint, techniques of variation are explored in each movement, all of which are based on the same theme. The improvised middle movement is based on original programs by the composer which explore ways of varying a pre-composed melody in real-time with the batons. Ideally, the radio-baton and flutist would improvise together. This was realized in a recent performance at the Palo Alto Cultural Center in December 1998, as part of a NACUSA concert. The piece was premiered in San Diego on October 10, 1997 at the Fourth Annual International New Music Festival, sponsored by the University of San Diego. At its premier, the radio-batonist performed a solo improvisation.

    Joanne D. Carey is a visiting composer at CCRMA.

### Chris Chafe

- *Whirlwind I and II* (1998)

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

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

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

- *Push Pull* (1995)

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

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

- *Between the Sheets*

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

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

**Ching-Wen Chao**

- *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.

**Janet Dunbar**

- *Songs of the Sea* (1997-8) for soprano, performance poet, keyboard, guitar, cello and CD

  A DMA final project in honor of the Internationational Year of the Ocean (1998) and the Yemaya, Mother of the Sea.

  *Songs of the Sea* is a cycle in ten sections with a mutable form which reflects the flexibility of the water element and having a variable length depending on which of the ten sections are performed. It is the last in a series of aural environments for a poet/photographer's mixed media installation. For this particular project, there are five sections of poetic text which are dramatically read over collages of sampled environmental sounds, algorithmically-generated sections seeded with Indian musical motives and coded with Common Music/Stella, and electric guitar improvisation which has been signal processed. Interspersed between these sections of performance poetry with recorded backgrounds, each portion of text is also set for soprano over written compositional material based on jazz chord progressions for synthesizer or piano, electric or acoustic guitar, and celletto or cello.

**Nicky Hind**

- *Cosmos* (1997)

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

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

**David A. Jaffe**

- *OTHER WORLDS: An homage to Carl Sagan*

  A concerto for Zeta electric/MIDI violin and symphonic band

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

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

- *The Seven Wonders of the Ancient World*

  A seventy-minute concerto in seven ten-minute movements for Boie-Mathews Radio Drum-controlled Disklavier.

  Instrumentation: Grand piano and ensemble of plucked string and percussion instruments: mandolin, guitar, harp, harpsichord, bass, 2 percussionists, harmonium.

  The piece recieved its world premiere by the San Francisco Contemporary Music Players in February, 1998 at the Yerba Buena Theatre in San Francisco. The San Francisco Chronicle described it as "a splendidly kaleidoscopic series of sketches, by turns exuberant, contemplative and austere." Work on the piece was supported in part by a Collaborative Composer Fellowship from the National Endowment for the Arts and the Banff Centre for the Arts, Canada. *The Seven Wonders of the Ancient World* was released on CD in 1996 on the Well-Tempered Productions label and was given an A+ rating by Audio magazine.

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

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

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

### Tobias Kunze

- *Protozoo* (1997)

  Live Stereo Sound Processing WP: CCRMA 11th Annual Industrial Affiliates Meeting, May 21-23, 1997, Stanford.

  *Protozoo* is a real-time, generative composition that creates a sequential, variative form solely as a result of a system of few basic audio processing operations. As such, it forms an acoustical analogon to dynamical systems often found in phenomena like chemical reactions, population growth, or models of processes in ecosystems. The listener is presented with a "zoo" of acoustical pre- and near-"life forms", simplistic, yet complex organisms, some of which develop to be more stable than others. Biological concepts like activation, inhibition, growth and death, transformation, digestion, inheritance and evolution come to mind and are helpful for the understanding of the composition.

  Realizations of Protozoo may be either in the form of a sound installation, an interactive instrument (using MIDI controllers), or as an effects processor.

  See Also: `http://www-ccrma.stanford.edu/~tkunze/mus`

### Jun Kim

- *Eum-Yang* (1998)

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

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

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

### Seungyon-Seny Lee

- *Chuk-won* (1998)

  *Chuk-won* is based on Samul nori which is a traditional from of Korean percussion music. Samul means "four things" in English and nori means "performing". The ensemble's members consist of two skins and two metals. The instruments symbolize earth (skins) and the heavens (metal). The instruments are identified with a constantly changing natural world. The metal instruments represent (1) Spring/lightening/thunder and (2)Summer/wind. The skin instruments represent (1) Autumn/rain and (2)Winter/clouds. It is said that if people play on these four instruments together, the resulting vibrations will harmonize earth and heaven into one universe. Sounds for this piece originate from recordings of skin and metal instruments used in the performance of Samul nori. During this performance, three video projectors display images that metaphorically combine with the music to reflect on the unity of creation. This piece forms the third part of a four-movement composition entitled Chuckwon which roughly translates as "invocation". This movement consists of electronic sounds only – other movements include a percussion quartet as well.

### Fernando Lopez Lezcano

- *iICEsCcRrEeAaMm*

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

- *House of Mirrors*

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

  *House of Mirrors* is an improvisational tour through a musical form and a four channel sound environment created by the composer/performer Fernando Lopez-Lezcano. The sound of doors opening and closing define the transitions between rooms, corridors and open spaces, where 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

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

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

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

**Juan Carlos Pampin**

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

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

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

- *Interstices* (1997) for string quartet

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

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

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

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

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

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

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

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

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

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

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

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

### Bob Sturm

- *Resurrection* (1998)

  This seven minute composition was generated from a one minute digital recording of improvised piano. Using Bill Schottstaedt's snd program in the linux environment, I stretched and pitch shifted the sample many times using granular synthesis until I arrived at a sound much more timbrally rich than the original solo piano.

### Kotoka Suzuki

- *Eclipse* (1996)

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

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

**Sean Varah**

- *Outside the Box* (1998)

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

- *Borderline*

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

- *Slipping Image*

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

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

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

**Marek Zoffaj**

- *In Principio Erat Verbum* (1998)

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

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

# 6 Research Activities

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

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

## 6.1 Computer Music Hardware and Software

### 6.1.1 The MusiCloth Project

**Lonny Chu**

The MusiCloth project is a study in the design and implementation of a performance environment for computer music that utilizes a graphical display along with a physical interface. The conceptual model for the MusiCloth is a large tapestry which the performer manipulates through large hand and arm motions and which produces MIDI output based on the performer's actions. Ultimately, the visual display should be implemented on a large, high-definition display so that the performer can stand before it, as if standing in front of a tapestry. The performer would then manipulate areas of the tapestry through hand and arm motions. This design would allow for both large, sweeping motions in addition to smaller, more precise control over smaller sections of the display. As a measure of flexibility, the graphical design of the display, along with its corresponding mappings to performance input and MIDI output, can be implemented using custom-designed overlays created by the composer. Currently, this project exists as a simple prototype to be run on a Power Macintosh G3. Eventually, however, the project should be ported to a large display system such as the Information Mural in the Stanford Computer Science department.

### 6.1.2 The CCRMA Music Kit and DSP Tools Distribution

**David Jaffe and Julius Smith**

Work is in progress by Stephen Brandon at the University of Glasgow to port the Music Kit to OPENSTEP. Leigh Smith is working on a port of MusicKit to Apple's PPC/OS X Server.

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

Release 4.2 is an incremental release with several significant additions:

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

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

- UPGRADED INTEL-BASED HARDWARE SUPPORT

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

- MOST RECENT APPLICATIONS

  Two Music Kit applications of note are available separately:

  - *Sequence*, a Music Kit Sequencer developed by Pinnacle Research. The new Sequence 0.9.85 release is available free from the CCRMA ftp server (`ftp://ccrma-ftp.stanford.edu/pub/-NeXT/Sequence.9.85.tar.Z`). This is an updated version released in 1998.
  - *SynthBuilder*, a synthesis instrument design and performance tool. SynthBuilder was the Grand Prize winner of the Second Annual International Music Software Competition at Bourges. It was developed by Stanford University's Sondius program, and is now being supported and further developed by Staccato Systems Inc. The NEXTSTEP version, including a free license authorization code, is available from `http://www.StaccatoSys.com` or `ftp://ftp.StaccatoSys.com`. Staccato Systems is also porting SynthBuilder to Windows 95, using the host CPU to do synthesis.

- RECENT UNIT GENERATORS

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

- OTHER RECENT FEATURES

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

### Other Music Kit News

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

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

### Music Kit Background

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

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

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

### 6.1.3 Mi_D

**Tobias Kunze**

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

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

### 6.1.4 PadMaster, an Interactive Performance Environment. Algorithms and Alternative Controllers

**Fernando Lopez Lezcano**

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

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

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

### 6.1.5 A Dynamic Spatial Sound Movement Toolkit

**Fernando Lopez Lezcano**

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

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

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

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

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

**Fernando Lopez Lezcano**

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

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

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

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

**Juan Pampin**

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

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

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

**Juan Pampin**

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

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

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

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

### 6.1.9 Computer-based implementation of Karlheinz Stockhausen's piece *Mantra*

**Juan Pampin**

Karlheinz Stockhausen's piece *Mantra* (1970), for two pianos and live-electronics, marked an important point in terms of real-time electronic music. Stockhausen's piece presents a whole network of interactions both in terms of instrumental actions and sound processing. The performers are required to control not only the intricate interplay between the two instruments but also to control the way the sound of their pianos is transformed by means of ring modulation. A noticeable gap in terms of "musical" interpretation arises here; while the players can control to a great extent the piano gestures carefully notated by the composer on the score; adjusting the parameters of ring modulation using a "dial" provided with the original analog equipment (designed by Stockhausen back in the 1970) becomes really awkward and complicated for them. The motivation for this project was to create a new interface for the dynamic control of the ring-modulators aiming both to keep the expression of the original setup, that obviously represents an important part of the piece (i.e. the "continuous" character of the dial, the grid of fixed frequencies/pitches, etc.), and to create an homogeneous interface for the pianists. The project was achieved in four stages:

1. Interface research In this stage the goal was to decide which interface was the most appropriate for the piece following the requests of a professional pianist (Tom Schultz). General questions of ergonomics were considered, specially regarding the use of keyboard interfaces and wheel controllers (as those available on commercial synthesizers).

2. Implementation of the live-electronics on the computer. In this stage the original analog sound processing modules were modeled in the computer using the CLM programming language. Some new capabilities were incorporated to the original model such as dc-blocking filters and low level controls.

3. Interface design. Based on the results of stage 1 the interface chosen for the frequency control of the ring-modulators was a MIDI keyboard synthesizer: Yamaha Sy77. This synthesizer allows for a multidimensional control of parameters through its keyboard and controllers, that can be easily mapped to the computer via MIDI. In the computer the controllers are scaled into the proper ranges by software, some of them used for coarse frequency changes (i.e. modulation wheel) and others for fine micro-tonal adjustments (i.e. dial). The keyboard note-on information is translated into tempered frequency values and velocity is mapped to portamento timing between frequencies, introducing an expressive dimension to the modulation changes.

4. Final software prototype design. The final prototype was implemented on an SGI computer running Common Lisp Music under Allegro Common Lisp 4.3. The program integrates a MIDI processing module (issued from stage 3) and a sound processing module that performs filtering and ring modulation (issued from stage 2) in two parallel channels (one for each piano). All controllers available on the Sy77 are accessible from the computer and can be mapped to any control parameter of the algorithms, allowing for a flexible design of the interface that can be different for each pianist. Actually, during the rehearsals of the piece (performed by Tom Schultz and Joan Nagano at Stanford in 1998) we had to adjust controller ranges and even change controllers on the fly by request of the players, trying to adjust the interfaces ergonomically at their request much as we could (for instance, Ms. Nagano's arm was too short to reach the modulation wheel of the synthesizer in time for during an intricate passage, after trying different solutions we set up things to have she playing on one of the frontal sliders closer to the piano keyboard.)

Conclusions: This computer implementation of Mantra not only opens the door for more performances of the piece without depending on its original analog gear (there are just a few working analog units that can be rented from the composer's editor), but it also allows for a new musical interpretation of the piece. The sound processing parameters are controlled from an homogeneous user interface that allows the pianists to "play" the modulation frequencies as notes on a keyboard and use wheels and sliders for coarse and fine tuning. Taking advantage of the digital implementation of the sound processing modules new features such as the dc-blocking filters were incorporated helping for better sonic results. Using the MIDI protocol new expression subtleties were introduced, expanding further more the musical interaction of the piece and integrating sound processing controls with the piano gestures.

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

**Juan Pampin**

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

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

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

#### 6.1.11 Stanford Computer-Music Packages for Mathematica

**Craig Stuart Sapp**

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

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

### 6.1.12 Graphical Additive Synthesis

**Craig Stuart Sapp**

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

### 6.1.13 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. STK currently works on SGI (Irix), Linux, NeXTStep, and Windows computer platforms. Oh, and it's free for non-commercial use. So, the code you write using STK actually has some chance of working in another 5-10 years. The only parts of STK that are platform-dependent concern real-time sound output and real-time MIDI input ... but we've taken care of that for you. The interface for MIDI input and the Tcl/Tk GUIs is the same, so it's easy to voice and experiment in real time using either the GUIs or MIDI.

STK is not a fancy graphical user interface (GUI). Why should we waste hundreds of hours making platform-dependent code, just so you can drag a box around with a mouse or view a sound in a display window? We would rather piggy-back off the extensive efforts of others. STK can generate simultaneous .snd, .wav, and .mat output soundfile formats (beside realtime sound output), so you can view your results using one of the numerous sound/signal analysis tools already available over the WWW (e.g. Snd, Cool Edit, Matlab). For those instances where a 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 the toolkit 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 SGIs, added realtime capabilities, and greatly expanded the synthesis techniques available. With the help of Bill Putnam, Perry also made a basic port of STK to Windows95. Gary Scavone began using STK extensively in the summer of 1997 and completed a full port of STK to Linux early in 1998. He finished the fully compatable Windows port (using Direct Sound API) in June 1998. With the release of version 2.0, STK offers unified functionality and performance on all platforms (except no realtime capabilities under NeXTStep).

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

**References**


- 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.

### 6.1.14 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.tar.gz, pub/Lisp/cmn.tar.gz, and pub/Lisp/snd.tar.gz.

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

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

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

### 6.1.16 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

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

was implemented at the Institut fr Musik und Akustik at the Zentrum fr Kunst und Medientechnologie in Karlsruhe, Germany, where the author worked for five years. Common Music continues to evolve today at the University of Illinois at Urbana-Champaign, where the author is now a professor of music composition. In 1996 Common Music received First Prize in the computer-assisted composition category at the 1er Concours International de Logiciels Musicaux in Bourges, France.

**Implementation**

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

**Synthesis Control**

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

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

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

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

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

**Contact**

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

## 6.2 Physical Modeling and Digital Signal Processing

### 6.2.1 Physical Modeling of Brasses

**David Berners**

One of the difficulties in building waveguide models of brasses and winds is that we do not know how to find the round-trip filtering in a flaring horn without actually making an acoustic measurement. Ideally, we would like to be able to compute the loop filter directly from the physical dimensions of the horn.

While significant work has been done along these lines (Causse et al. [1], Plitnik and Strong [2], Benade [3]), a complete and accurate theory is not yet available.

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

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

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### 6.2.2 Numerical Integration of Partial Differential Equations

**Stefan Bilbao**

This work focuses on a numerical integration method for partial differential equations (PDEs) which is an outgrowth of Wave Digital Filtering (WDF), a well-known digital filter design technique. The idea is, as in the lumped case, to map a Kirchoff circuit to a signal flow diagram in such a way that the energetic properties of the various circuit elements are preserved. The method can be extended to distributed systems as well. The chief benefit of this method, which has been around for some ten years now, is its guaranteed stability under very general conditions. Applications are to modelling distributed acoustic, electromagnetic and even highly nonlinear fluid dynamics phenomena. Current work is concerned with, in particular, a WDF version of the PML (perfectly matched layer) used for unbounded domain problems in electromagnetics, flux-splitting in the WDF framework, incorporating the entropy variable formulation of fluid dynamics, higher-order accurate discretization formulae which preserve passivity, and other projects.

### 6.2.3 Vicarious Synthesizers: Listening for Timbre

**Chris Chafe**

The timbre of a digitally synthesized musical sound is usually determined by a group of controls, as for example in physical models of bowed strings (bow force / velocity / position) or sung vowels (complex vocal tract shape / glottal source) and in models using frequency modulation (modulation index / oscillator tuning ratios). In this work which concentrates on the bowed string physical model, possible tone qualities are arrayed in a two-dimensional matrix whose axes (bow force / velocity) represent two of the principle timbral determinants of the synthesis method. Expressive control of timbre in real-time is achieved by navigating the space with a force-feedback pointing device, allowing the musician to feel as well as hear timbral change. Timbral features are displayed kinesthetically as variations in the graph surface. Locations of particular bowed timbres and nearby qualities in their environs are easily learned along with musically important trajectories. The representation provides a window on bowing technique in digitized performances by tracking spectral matches between the recording and matrix.

### 6.2.4 Synthesis of Transients in Classical Guitar Sounds

**Cem Duruoz**

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

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

**Shyh-Kang Jeng**

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

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

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

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### 6.2.6 Scalable Audio Models for Data Compression and Modifications

**Scott Levine**

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

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

See Also: http://www-ccrma.stanford.edu/~scottl/thesis.html

### 6.2.7 Articulatory Singing Voice Synthesis

**Hui-Ling Lu**

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

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

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

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

### 6.2.8 Perceptual Audio Coding Based on the Sinusoidal Transform

**Juan Pampin and Guillermo Garcia**

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

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

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

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

**Craig Stuart Sapp**

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

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

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

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

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

**Gary P. Scavone**

The modeling of musical instruments using digital waveguide techniques has proven to be both an accurate and efficient technique for synthesis. Because such models are based on physical descriptions, they further provide a useful tool for acoustical explorations and research. Models of wind instrument air columns have reached a high level of development. An accurate and efficient means for modeling woodwind toneholes was described in [Scavone and Cook, 1998].

Recent work has focused on the modeling of three-dimensional sound radiation from woodwind and brass instruments. Past wind instrument models have been primarily concerned with the acoustic behavior internal to the air column, and have neglected the direction-dependent sound radiation characteristics of the instrument being modeled. That is, most current models implement either a virtual pressure pickup somewhere inside the instrument air column or, for a slight improvement, apply a simple high-pass filter to the internal sound pressure to mimick transmission through a bell. While this last technique offers a fairly accurate representation of the sound transmitted in a one-dimensional sense out of the bell, it does not offer any angular dependency with respect to arbitrary pickup points external to the instrument.

Currently available acoustic theory with regard to sound radiation from ducts and holes can be implemented in the digital waveguide context using properly designed digital filters. In general, first- and second-order digital filters are sufficient to accurately model the angular dependence of sound radiation from flanged and unflanged pipes and various brass bell contours. In order to test the accuracy of such models, frequency-domain polar radiation data was recently measured in an anechoic chamber at the Institut de Recherche et Coordination Acoustique/Musique (IRCAM). Further, an experimental setup to obtain time-domain, impulse-response data through maximum-length sequence techniques is currently under development.

While the addition of a single second-order digital filter represents a fairly insignificant increase in computational complexity in brass instrument waveguide models, the implementation of individual filters for each tonehole of a woodwind instrument model quickly becomes prohibitive in most synthesis environments. This work is thus addressing the issue of model complexity. A simplified system appropriate for real-time implementation is being developed that allows continuous pickup movement within a 3D space. This synthesis model is then combined with a 3D Virtual Reality Modeling Language (VRML) model to offer corresponding realtime visual and aural response.

All realtime synthesis development is being performed using The Synthesis ToolKit (STK).

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### 6.2.11 FFT-Based DSP and Spectral Modeling Synthesis

**Julius Smith**

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

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

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

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

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

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### 6.2.12 Digital Waveguide Modeling of Acoustic Systems

**Julius Smith**

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

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

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

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

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### 6.2.13 Signal Processing Algorithm Design Stressing Efficiency and Simplicity of Control

**Timothy Stilson**

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

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

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

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

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

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

### 6.2.14 Antialiasing for Nonlinearities: Acoustic Modeling and Synthesis Applications

**Harvey Thornburg**

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

### 6.2.15 A Speech Feature Based on Bark Frequency Warping – The Non-uniform Linear Prediction (NLP) Cepstrum

**Yoon Kim**

In statistically based speech recognition systems, choosing a feature that captures the essential linguistic properties of speech while suppressing other acoustic details is crucial. This could be more appreciated by the fact that the performance of the recognition system is bounded by the amount of linguistically-relevant information extracted from the raw speech waveform. Information lost at the feature extraction stage can never be recovered during the recognition process.

Some researchers have tried to convey the perceptual importance in speech features by warping the spectrum to resemble the auditory spectrum. One example is the mel cepstrum (Davis, 1980), where a filterbank that has bandwidths resembling the critical bands of human hearing is used to obtain a warped spectrum. Another is the Perceptual Linear Prediction (PLP) method proposed by Hermansky (Hermansky, 1990), where a filterbank similar to the mel filterbank is used to warp the spectrum, followed by perceptually motivated scaling and compression of the spectrum. Low-order all-pole modeling is then performed to estimate the smooth envelope of the modified spectrum.

While the PLP provides a good representation of the speech waveform, it has some disadvantages that should be pointed out. First, since the PLP method relies on obtaining the FFT spectrum before the warping, its ability to model peaks of the speech spectrum – formants – depends on the characteristics of the harmonic peaks for vowels. This could hinder the process of modeling formants of female speech through filterbank analysis, since there are fewer harmonic peaks under a formant region than in the male case. Second, various processing schemes (e.g. Bark-scale transformation, equal-loudness weighting, cubic-root compression) require memory, table-lookup procedure and/or interpolation, which might be computationally inefficient.

We propose a new method of obtaining parameters from speech that is based on frequency warping of the vocal-tract spectrum, rather than the FFT spectrum. The Bark Bilinear Transform (BBT) (Smith, 1995) is first applied on a uniform frequency grid to generate a grid that incorporates the non-uniform resolution properties of the human ear. Frequency warping is performed by taking the non-uniform DFT (NDFT) of the impulse response related to the vocal-tract transfer function using the warped grid. The warped spectrum is then modeled by low-order Linear Prediction (LP), which provides a good estimate of the spectral envelope, especially near peaks. This results in features that effectively model the warped peaks of the vocal-tract spectrum, which are considered to be perceptually important. Results of vowel classification experiments show that the proposed feature effectively captures linguistic information while suppressing speaker-dependent information due to different acoustic characteristics across speakers.

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## 6.3 Controllers for Computers and Musical Instruments

### 6.3.1 Real-time Controllers for Physical Models

**Chris Chafe**

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

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

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

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

### 6.3.2 The Touchback Keyboard

**Brent Gillespie**

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

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

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

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

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

### 6.3.3 Updates on the Radio-Baton Program

**Max V. Mathews**

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

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

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

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

### 6.3.4 Haptic User Interfaces for the Blind

**Sile O'Modhrain and Brent Gillespie**

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

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

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

**Craig Stuart Sapp**

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

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

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

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

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

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

**Gary P. Scavone**

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

- *The Holey Controller*

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

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

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

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

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

## 6.4 Psychoacoustics and Cognitive Psychology

### 6.4.1 Neural Network Models of Musical Cognitive Activities

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

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

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

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

### 6.4.2 New Text on Psychoacoustics

**Perry Cook**

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

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

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

### 6.4.3 Absolute Pitch, Absolute Tempo, Absolute Loudness

**Daniel Levitin**

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

More specifically, my current research projects include work on:

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

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

## 6.5 Machine Recognition in Music

### 6.5.1 Statistical Pattern Recognition for Prediction of Solo Piano Performance

**Chris Chafe**

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

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

### 6.5.2 Optical Recognition of Printed Music: A New Approach

**Walter Hewlett**

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

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

## 6.6 Historical Aspects of Computer Music

### 6.6.1 Impact of MIDI on Electroacoustic Art Music

**Alex Lane Igoudin**

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

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

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

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

However, one can see the emergence of new methods, new practices and new performance solutions that were not present in the pre-MIDI era.

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

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

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

## 6.7 Computer Assisted Music and Acoustics Research

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

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

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

#### Teaching and Research Functions

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

#### Databases

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

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

MuseData(TM) databases are device-independent. That is, all of the data are stored in ASCII representations. The database codes, file structures, and operating software have been developed by Walter B. Hewlett. Technical information is available in a separate publication from CCARH. To facilitate applications, works represented in MuseData(TM) code are translated into other application-specific codes. At the present time the application-specific codes supported are MIDI, SCORE, DARMS, and Kern.

MIDI supports sound and sequencer applications in multiple platforms. SCORE and DARMS support printing applications on DOS microcomputers. Kern supports analytical applications under UNIX and UNIX simulation software (such as the MKS Toolkit, running under DOS). Other openly documented codes may be supported in the future.

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

**Publications**

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

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

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

**Invited visitors**

- David Huron, developer of the Humdrum Toolkit for Musical Analysis and newly appointed professor of Music and Cognition at the Ohio State University, was in residence in 1996-97. His stay enabled him to complete a user guide for the Humdrum community, to offer a summer course on Húmdrum, to develop a database of musical themes searchable on the World-Wide Web, to further develop conversion software from the MuseData format to the Kern format used by Humdrum, and to participate in various teaching and research activities.
- Andreas Kornstaedt, a graduate student in software engineering at the University of Hamburg (Germany), spent 1996-97 in residence at CCARH. He was involved in developing a user interface for thematic searches, in developing CCARH's Web capabilities, and in creating an enriched Kern format that preserves page-description information used by the SCORE notation program. Mr. Kornstaedt wrote a conversion program from the SCORE binary file format to this enriched version of Kern and assisted Dr. Huron in many of his activities.
- Douglas R. Hofstadter, the well-known author of Goedel, Escher, Bach and numerous other books concerned with issues of cognition, organized a lecture series on computer challenges to human creativity for the Autumn quarter of 1997-98. Prof. Hofstadter was on leave from Indiana University.

### 6.7.2 The Musical Acoustics Research Library

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

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

# 7 Recordings

Recordings of works realized at CCRMA include the following:

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

- *Electroacoustic Music II.* Music by Berger, Child, Dashow, Duesenberry, Shapiro (Jonathan Berger: "An Island of Tears"), Neuma 450-73-[J].
- Michael McNabb. *Computer Music.* Digitally mastered LP. McNabb – "Dreamsong," "Love in the Asylum," "Orbital View," (LP out of print) - CD now available as WER-2020-2, Wergo, Germany, 1994, Harmonia Mundi, Distributors.
- Michael McNabb. *Invisible Cities.* Digital compact disk, WER-2015-50, Wergo, Germany, 1988, Harmonia Mundi, Distributors.
- *Musica Maximalista - Maximal Music Vol. 2,* – digital compact disk containing Celso Aguiar – "Piece of Mind", CD MM-002, Studio PANorama, Brazil, 1996.
- *Musica Maximalista - Maximal Music Vol. 3,* – digital compact disk containing Celso Aguiar – "All blue, I write with a blue pencil, on a blue sky", CD MM-003, Studio PANaroma, Brazil, 1997. CD of the II International Electroacoustic Music Competition of So Paulo.
- *Night Chains.* (Composers Recordings Inc - Emergency Music Series, CRI CD681). Jeffrey Krieger, electronic cello (Jonathan Berger: "The Lead Plates of the Rom Press").
- *II SBCM (II Brazilian Symposium on Computers and Music),* – digital compact disk containing Celso Aguiar – "Piece of Mind", DISC MFG., INC., BHS1046, Brazil, 1996.
- *The Science of Musical Sound - Musical Examples.* Cassette tape produced by Jan Mattox for CCRMA to accompany the book *The Science of Musical Sound,* J. R. Pierce, Scientific American, 1985.
- *Unknown Public (04): Musical Machinery,* – digital compact disk containing Nicky Hind – "Rain", Unknown Public, UPCD04, United Kingdom, 1994.

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

# 8 Publications

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


Berger, J. (1998). Connected to what? *Journal of the College Music Society*, (1). Plenary address, 1997.

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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).

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