CENTER FOR COMPUTER RESEARCH IN MUSIC AND ACOUSTICS

DEPARTMENT OF MUSIC, STANFORD UNIVERSITY

REPORT NO. STAN-M-98

# CCRMA REPORT

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EDITED BY

ALEX IGOUDIN

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# Chapter 1

# General Information

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

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

The CCRMA community consists of administrative and technical staff, faculty, research associates, graduate research assistants, graduate and undergraduate students, visiting scholars and composers, and industrial associates. Departments actively represented at CCRMA include Music, Electrical Engineering, Mechanical Engineering, Computer Science, 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, and the Journal of the Acoustical Society of America. 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, E-mu, Dynacord, 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.

# Chapter 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".

### 2.0.1 Staff and Faculty

| Login | Name | Position |
|---|---|---|
| CC | Chris Chafe | Assoc. Prof. of Music, Music Dept. Chair |
| JEG | Johannes Goebel | Technical Director (ending 7/31/96) |
| NANDO | Fernando Lopez-Lezcano | System Administrator / Lecturer |
| JAY | Jay Kadis | Audio Engineer / Lecturer |
| HMK | Heidi Kugler | Secretary |
| MVM | Max V. Mathews | Professor of Music (Research) |
| BRG | Jonathan Berger | Associate Professor of Music |
| JOS | Julius O. Smith III | Associate Professor, Music and Electrical Engineering |
| JC | John Chowning | Professor of Music, Emeritus |
| LCS | Leland Smith | Professor of Music, Emeritus |
| JRP | John R. Pierce | Visiting Professor of Music, Emeritus |
| ROGER | Roger N. Shepard | Professor of Psychology, Emeritus |
| EDS | Earl Schubert | Professor of Speech and Hearing, Emeritus |
| JDH | Jonathan Harvey | Professor of Music |
| DBS | David Soley | Assistant Professor of Music |
| ESF | Eleanor Selfridge-Field | Consulting Professor of Music |
| n/a | Walter Hewlett | Consulting Professor of Music |
| BAU | Marcia Bauman | Research Associate, IDEAMA Archive |
| BIL | William Schottstaedt | Research Associate |

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

### 2.0.2 Graduate Students

| Login | Name | Position |
|---|---|---|
| CELSO | Celso Aguiar | DMA (Doctor of Musical Arts) |
| GCARLO | Giancarlo Aquilanti | DMA (Doctor of Musical Arts) |
| JAN | Jan Chomyszyn | PhD Computer-Based Music and Acoustics |
| DATTORRO | Jon Dattorro | PhD Electrical Engineering |
| KUI | Kui Dong | DMA (Doctor of Musical Arts) |
| JMD | Janet Dunbar | DMA (Doctor of Musical Arts) |
| MICHAEL | Michael Edwards | MA, DMA (Doctor of Musical Arts) |
| DURUOZ | Cem Duruoz | PhD Electrical Engineering, MA Music |
| RJFLECK | Robert Fleck | PhD Computer-Based Music and Acoustics |
| BRENT | Brent Gillespie | PhD Mechanical Engineering |
| n/a | Edward Gross | PhD Civil Engineering |
| NICKY | Nicholas Hind | DMA (Doctor of Musical Arts) |
| ALEDIN | Alex Igoudin | PhD Computer-Based Music and Acoustics |
| YOONIE | Yoon Kim | PhD Electrical Engineering |
| JUNKIM | Jun Kim | DMA (Doctor of Musical Arts) |
| TKUNZE | Tobias Kunze | DMA (Doctor of Musical Arts) |
| n/a | Jonathan Lederman | PhD Computer-Based Music and Acoustics |
| BOBBY | Bobby Lombardi | DMA (Doctor of Musical Arts) |
| JMCGINN | John McGinn | DMA (Doctor of Musical Arts) |
| EIG | Enrique I. Moreno | PhD Computer-Based Music and Acoustics |
| NORTON | Jonathan Norton | PhD Computer-Based Music and Acoustics |
| SILE | Sile O'Modhrain | PhD Computer-Based Music and Acoustics |
| JUAN | Juan Pampin | PhD Computer-Based Music and Acoustics |
| CRAIG | Craig Sapp | PhD Computer-Based Music and Acoustics |
| GARY | Gary P. Scavone | PhD Computer-Based Music and Acoustics |
| KOTOKA | Kotoka Suzuki | DMA (Doctor of Musical Arts) |
| WISHBON | Steven Trautmann | DMA (Doctor of Musical Arts) |
| SAVD | Scott Van Duyne | PhD Computer-Based Music and Acoustics |
| LEIGH | Leigh VanHandel | PhD Computer-Based Music and Acoustics |
| PVH | Paul Von Hippel | PhD Computer-Based Music and Acoustics |
| ANNW | Ann Williamson | PhD Computer Science |

### 2.0.3 Student Research Assistants

| Login | Name | Program |
|---|---|---|
| DPBERNER | David P. Berners | PhD EE |
| BILBAO | Stephan Bilbao | PhD EE |
| YYC | Yi-Yin Chen | PhD EE |
| SCOTTL | Scott Levine | PhD EE |
| PUTNAM | William Putnam | PhD EE |
| STILTI | Timothy Stilson | PhD EE |

### 2.0.4 Undergraduates including MST (Music, Science and Technology) Majors

| Login | Name | Program |
|---|---|---|
| YASMIN | Yasmin Craig | MST |
| CMGOODAN | Charles Goodan | MST |
| CURTIS | Ethan Eldridge | MST |
| AQUEELAH | Aqueelah Haqq | MST |
| SCOTT | Scott McKissen | BA Psychology |
| MEJANE | Jane Rivera | MST |
| ISSAC | Issac Roth | Multimedia Performance Design and Technology |
| JAISCNCE | Jaijae Soto | MST |
| TRACE | Trace Wax | MST, Computer Science |

### 2.0.5 Visiting Scholars

| Login | Name | Home Affiliation |
|---|---|---|
| MAB | Marina Bosi | Research Engineer, Dolby Labs |
| SBRAND | Steffen Brandorff | Researcher, Denmark |
| JDC | Joanne Carey | Composer, USA |
| PRC | Perry R. Cook | Assistant Prof., Computer Science and Music, Princeton |
| n/a | Richard Festinger | Composer, USA |
| FORNARI | Jose Fornari | PhD Student, University of Campinas, Brazil |
| PHILIP | Philip Goyal | Researcher, UK |
| DHURON | David Huron | Researcher, Canada |
| DAJ | David Jaffe | Composer, Software Research Engineer, USA |
| PEER | Peer Landa | Composer, Norway |
| LUKAS | Lukas Ligeti | Composer, Austria |
| n/a | Jan Mattox | Researcher , USA |
| JOSE | Jose Montalvo | Composer, Puerto Rico |
| DEX | Dexter Morrill | Professor, Composition, Colgate University |
| MULLER | Carl Muller | Researcher, USA |
| n/a | Joe O'Keefe | Researcher/Composer, USA |
| PASI | Fiammetta Pasi | Composer, Italy |
| DOUDOU | Xavier Perret | MS Student, Ecole Polytechnique, Paris France |
| NICK | Nick Porcaro | Software Research Engineer, Stanford OTL |
| ROC | Davide Rocchesso | PhD Student, Universita di Padova, Italy |
| GPS | Pat Scandalis | Software/Hardware Research Engineer, Stanford OTL |
| STEPHEN | Stephen Schwanauer | Visiting Scholar, Lecturer |
| XJS | Xavier Serra | IUA - Phonos, Universitat Pompeu Fabra, Barcelona, Spain |
| MALCOLM | Malcolm Slaney | Visiting Scholar, Lecturer |
| HKT | Rick Taube | Assistant Professor: University of Illinois |
| MARCO | Marco Trevisani | Composer, Verona, Italy |
| PATTE | Patte Wood | ICMA, USA |

# Chapter 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 with andadjoining 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 hardware environment consists of an Ethernet network of workstations that include NeXT machines, PC's running NEXTSTEP and Linux, SGI's and Macintosh computers. Digital signal processing is possible on all workstations, both through the use of the main processor and built in (in the case of NeXT) or add on (in the case of PC's) Motorola 56001 DSP processors. The Macintosh systems also provide DSP via Digidesign Sound Accelerator boards. MIDI input and output are supported on all platforms. Four channel soundfile playback is supported on the NeXT's through custom built 4 channel D/A converters. Digital audio processors include a Studer-Editech Dyaxis II system which can convert all popular digital audio formats as well as store and edit audio digitally, a Digidesign Pro-Tools system with a Sony CD-R drive, several Singular Solutions analog and digital audio input systems for the NeXTs, and several Panasonic and Tascam DAT recorders. Text and graphics are handled by an HP 4c color scanner on the NeXT system and by two NeXT Laser printers and an AppleLaserwriter 630 Pro.

The MIDI-based systems include various Macintosh computers with Yamaha, Roland and Korg equipment including Yamaha DX, SY, TG and VL synthesizers, KX88 keyboard controller, Korg X3R, I3, WaveDrum and WaveStations, an E-mu Systems Emulator IV and ESI-32 sampler, and digital delays and reverberation. The mixing console is a Yamaha RM2408 in the MIDI Studio and an Allen and Heath GL2 in Studio E. Speakers are Ramsa WS-A200: stereo with subwoofer in MIDI studio and quad in Studio E. Other equipment available includes IVL pitch trackers, a Buchla Lightning MIDI controller, a Mathews Radio Drum controller, an Opcode Studio 5 MIDI patcher, 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, 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-235 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 room (Studio D) consists of a Dyaxis II digital editing system, Digidesign Sound Designer II system, Meyer 833 monitor speakers, a Yamaha DMP-7D digital mixer, Yamaha SPX-1000 processor, a Panasonic SV-3700 DAT recorder and Macintosh and NeXT computers. Digital and analog patchbays provide interconnection.

The CCRMA software has been developed over a twenty-year period, and consists of a vast set of programs and system tools for editing, viewing, synthesizing, and analyzing sound. Much of the software was originally written in SAIL, a sophisticated Algol-like language for use on the previous mainframe and has been ported to the new workstation environment and developed further. The programs currently in use include a comprehensive environment written in Common Lisp that includes Common Lisp Music (CLM) for music synthesis and signal processing and Common Music (CM) and STELLA for compositional programming. 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, and Smalltalk. A graphical environment for real-time DSP research, SynthBuilder[1], is currently under development. Of course there is a wide variety of public domain software installed on all workstations.

[1] http://www-leland.stanford.edu/group/OTL/SynthBuilder.html

# Chapter 4

# Academics

CCRMA is a part of the Music Department 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, i.e., Music, Computer Science, Electrical Engineering, Psychology, etc. Graduate degree programs offered in music are the MA in Musical Acoustics, Perception, and Synthesis, 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. The specialization in Music, Science and Technology 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.

### University Courses

For complete information on the following classes, please see Stanford Bulletin for the current academic year.

Courses offered at CCRMA for Stanford students include:

- **Music 15A. Topics in Interactive Computer-Music Performance.** For sophomores only. Real-time interactive performance for interested musicians combining composition, performance, MIDI instruments, and computer programming. Introduction to programming, composition of short pieces, moving beyond familiar styles. Prepares students for work in ensembles and CCRMA courses.

- **Music 120. Introduction to Composition and Programming using MIDI-Based Systems.** Composition projects demonstrate participant's own software for voicing and controlling MIDI synthesis.

- **Music 149. Instrumental Music with Electronics.** The link between "traditional" evaluation of instrumental, orchestral, and vocal music and the revolutionary world of the electronic studio occurs in works where the two are combined. The course focuses on such linking works, beginning with

Stockhausen's contributions and moving on to the products of IRCAM (Boulez, Murail, etc) and elsewhere.

- **Music 151. Psychophysics and Cognitive Psychology for Musicians.** Basic concepts and experiments relevant to use of sound, especially synthesized, in music. Introduction to elementary concepts; no previous background assumed. Listening to sound examples important. Emphasis on salience and importance of various auditory phenomena in music.
- **Music 154. History of Electroacoustic Music.** Survey of works and techniques.
- **Music 192. Theory and Practice of Recording**
  - **192A. Foundations of Sound Recording Technology.** Topics: elementary electronics, physics of transduction and magnetic recording of sound, acoustic measurement techniques, operation and maintenance of recording equipment, recording engineering principles.
  - **192B. Advanced Sound Recording Technology.** Topics: digital audio including current media, formats, editing software, and post-processing techniques. Also, microphone selection and placement, grounding and shielding techniques, noise reduction systems and advanced multi-track techniques.
  - **192C. Session Recording.** Independent engineering of recording sessions.
- **Music 220. Computer-Generated Music**
  - **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[1]. 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.**

[1] http://www-ccrma.stanford.edu/CCRMA/Courses/220b/

- **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[2]. 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 423. Digital Signal Processing Research Seminar.** Ongoing seminar for doctoral students pursuing research in DSP applied to music or audio.

### Workshops

CCRMA also offers a series of two-week summer workshops open to the participants outside of Stanford community. Courses offered in 1994-96 included the following:

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

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

- **Introduction to Psychoacoustics and Psychophysics: Audio and Haptic Components of Virtual Reality Design**

  This course introduces concepts and applies 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 are examined. During the course of the workshop, composers and virtual object designers study the perceptual sciences, such as acoustics, psychology, physics and physiology, to create virtual environments which are convincing upon hearing and touch. They learn to 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 are addressed. Various audio and haptic effects and illusions are demonstrated.

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

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

- **Introduction to Algorithmic Composition**

  This course introduces basic principles and techniques of algorithmic composition and covers such topics as object oriented music representation, chance composition, algorithmic description of musical processes, and musical pattern languages. Sound synthesis performed as course material include MIDI, the (realtime) Music Kit and (non-realtime) Common Lisp Music. The course is taught using the Common Music environment* on Mac and NeXT workstations. The labs are hands-on spectral and physical modeling using software such as SMS, MusicKit, SynthBuilder, and simple C-Code examples. The Yamaha synthesizers used in the course include the VL-1 and SY-77. All source code and documents from the workshop including the graphic interface are free to take.

- **Advanced Projects in Algorithmic Composition**

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

- **Computer-Assisted Research in Musicology**

  This course, offered in cooperation with the Center for Computer Assisted Research in the Humanities, provides a comprehensive introduction to computer-assisted research in musicology and ethnomusicology using the Humdrum Toolkit. Participants learn to manipulate computer-based scores, tablatures, and other documents in order to solve a wide variety of musicological problems. E.g., participants learn to characterize common patterns of orchestration in Beethoven symphonies, examine harmonic progressions in Bach chorale harmonizations, and investigate text/melody relationships in Gregorian chant. Thousands of full scores are made available for processing on-line - including repertoires from various cultures, periods, and genres. The course is of particular value to scholars contemplating graduate-level or advanced music 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.

- **Music Printing with Small Computers Using SCORE**

  This course covers the details of the use of the SCORE software program for the creation of publication-quality music typography on PC compatible computers. Emphasis is placed on individual projects.

- **Intensive Digital Signal Processing**

  This three-day spring workshop covers applications of the Fast Fourier Transform (FFT) arising in digital audio research. The main topics addressed are practical spectrum analysis using the FFT, sound synthesis by means of spectrum models, and signal processing using the FFT.

  Specific topics include FFT windows, spectrum analysis, FFT based convolution, and phase vocoders. Both the overlap-add and filterbank-summation interpretations of short-time Fourier processors are addressed. Additionally, applications such as audio compression, and time/compression and expansion are presented.

# Chapter 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 and Intel based PCs) running the NEXTSTEP operating system and the functionality of PLA exists now in the form of Common Music (CM) and STELLA (written in Common Lisp by Rick Taube), a software package that can write scores by listing parameters and their values, or by creating algorithms which then automatically determine any number of the parameters' values. Common Music (CM) can write scores in several different syntaxes (currently CLM, CMN, Music Kit, MIDI, CSound and Paul Lansky's real-time mixing program, RT). The scores can then be rendered on workstations using any of the target synthesis programs. For example, CLM (Common Lisp Music, written by Bill Schottstaedt) is a widely used and fast software synthesis and signal processing package that can make use of multiple Motorola 56001 DSPs or direct synthesis on faster workstations (including real time). CCRMA has also become the maintainer and distributor of NeXT's Music Kit, a real-time toolkit for computers running NEXTSTEP that merges the MIDI and real-time synthesis paradigms and can also be the target of Common Music generated scores.

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

and developmental research for all these systems include **MAX, LeLisp, Smalltalk, Common Lisp, DMIX** (a flexible compositional environment, written by Dan Oppenheim), **Objective C** and **C**.

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

## 5.2 Composers and Works

Recent compositional works realized at CCRMA include the following:

### Celso Aguiar

*Piece of Mind* and *Ayahuasca*

Both are tape pieces using spectral modeling (SMS), sampling and granular synthesis in CLM and CM's Lisp environment. *Piece of Mind* was awarded "Premio Sao Paulo '95", Brazil; recorded on a CD released by the II SBCM and CRCA (UCSD) in 1995.

### alt.music.out

*Wonderment in Eb* (1995)

Live interactive improvisatory piece by Nick Porcaro and David Rhoades, where the performers improvise in harmolodic manner over several phrases of jazz-based material. Running on several NeXT computers, the SynthBuilder application is used as a real-time effects processor and Physical Modeling (PM) synthesizer. Both the performers and a sound engineer have control over the effects. alt.music.out are: Emily Bezar - Processed vocals; Roberto DeHaven - Processed drums, saxophone; Scott Levine - Effects processing; Nick Porcaro - Processsed grand piano and PM piano; David Rhoades - Processed saxophone; Pat Scandalis - PM electric guitar, electric guitar; Tim Stilson - Effects mixing. Premiered at the CCRMA Annual Summer concert, July 1995.

### Joanne D. Carey

*Gracias*

The last interactive piece of a trilogy for soprano and radio-baton. *Gracias*, as well as its companions *La Soledad* (1992) and *Aqui* (1993), was inspired and influenced by Spanish Flamenco and indigenous South American music, and the later poetry of Chilean poet Pablo Neruda. The spirituality and humanity of this great poet continues to impress the composer deeply. In the process of blending Neruda's poetry with the rhythms, flourishes and instrumental sound of these Spanish and South American musical traditions, Joanne Carey drew mainly from strains of solitary meditation and deep sorrow buoyed by irrepressible exuberance and hope. The scores of the electronic accompaniments were created on a MacintoshIIfx using the DMIX composition program developed by Daniel Oppenheim. The sound material for these songs was generated on a Yamaha SY77. Most of the voices are presets, with the exception of the bell sounds and a couple of hybrid sounds that were constructed by the composer and a "sliding sigh" sound developed by Dr. Oppenheim.

The composition has been widely performed: San Jose Chamber Music Society, 1995; SEAMUS conference in Ithaca, New York, 1995; IBM Research Center, Yorktown, New York, 1995; International New Music Festival, San Diego, 1995; University of Maryland, Demo concert with Max Mathews, 1995; Radford, Virginia, Demo concert with Max Mathews, 1995; Peabody Conservatory, Baltimore, Maryland, Demo concert with Max Mathews, 1995; National Association of Teachers of Singing, Winter Vocal Symposium, 1996.

### Chris Chafe

*Push Pull* (1995)

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

evolve. The work is a study for a new piece for singer and computer sponsored by a grant from the National Endowment for the Arts.

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

### Kui Dong

*Flying Apples* (1994)

The contents of the piece concerns an unfinished childhood dream where, with the unlimited imagination of the child, a walk is taken through a colorful and unspoiled world. The piece was composed using DMIX, a newly developed software for Macintosh. It was programmed with extreme nesting patterns, forming a simple idea which then grows into a complex pattern. While composing, Kui Dong does not think excessively about tools and techniques, instead she listens for what best fits her overall concept for a piece of music, looking for the right color and shape for each sound. Purity and sincerity are truths that guide her. In Flying Apples she attempted to catch the transparent, brilliant stars falling from infinity.

Performed at Visual Symbols, San Jose; Stanford Univeristy; ICMC 1995, Banff, Canada; LIPM, Buenos Aires, Argentina.

*Cess, After Turandot* (1995-1996)

A chamber opera in three acts for eight voices, ten instruments, and a tape with a duration of approximately sixty minutes. To be premiered by Other Mind Festival in November, 1996.

The story is based on the famous play but goes beyond the original libretto by delving specifically into the themes of identity and desire. The Ice Princess, who is also the central figure in Puccini's opera, is named Cess and is an idolized underground nightclub performer. Part of her on and off stage attempt to thwart admirers is to offer the challenge of cracking the enigma of who she really is, with the risk that the wrong answer will bring death. A gangster, new to the area, takes up the challenge and through a series of dream-like realizations discovers that Cess is a Hermaphrodite. When he reveals her identity the crowd grows enraged. The mystique of their idol has been disclosed and they retaliate by savagely murdering the gangster.

Kui Dong is a winner of 1994 Alea III International composition Prize, Boston; 1990 National Art Song Competition; 1989 National Music and Dance Competition Beijing; received awards from 1995 ASCAP Grants to Young Composers Contest; 1995 Santa Clara Commissioning Award for Art; 1995 Djerassi Foundation for Art; 1993 Asia-Pacific National Fund and Stanford University.

### Cem Duruoz

*Cycles*

An interactive piece for Classical Guitar, NeXT (Physically Modeled SynthBuilder Flute), Macintosh (sequencer). Performed at Stanford University, February 4, 1996.

Cem Duruoz is a recepient of Stanford Student Soloist Award (1992); Winner, Turkish Classical Guitar Competition (1984), and Semi-finalist, Guitar Foundation of America International Competition, (1993, 1995).

### Michael Edwards

*Brownian Motion* (1994)

Composed for stereo tape using samples processed in CLM, and note lists generated by Common Music, mixed using Paul Lansky's RT Mixer app. Performed at Stanford University; Thessaloniki, Greece; Buenos Aires, Argentina, 1995, Belfast, Northern Ireland 1996.

*segmentation fault beta1.0* (1996)

For prepared piano and computer, real-time mixing software for the SGI computer. Incorporates sound-file-to-keyboard mapping that allows sounds to be triggered at the keyboard of the computer during live perormance. Premiered at Stanford University, 1996.

**Doug Fulton**

*Holding Betty under Water* (1995)

For computer-generated tape.

**Nicky Hind**

Recordings of a series of pieces for 3 female voices - *The Well,The Gentle*, and *The Joyous*, with tape, vibraphone, and harp respectively - are in progress for forthcoming CD release. An ongoing series of works for solo instrument and live electronics is also in progress - *Ripples* (guitar), and *Prisms* (piano) so far completed. Other activities include audio production of contemporary music e.g. the CDs Viola Pieces, and Minimalist Guitar Music.

*The Gentle*

The title refers to a character (or hexagram) from the I Ching (the Chinese Book of Changes), whose meaning is concerned with the way subtle forces–over a prolonged period–can often have a powerful and penetrating effect. The results of which, although "less noticeable than those won by surprise attack, are more enduring and more complete". Composed in 1991 (but extensively revised in 1995), *The Gentle* is the first of what became a series of three pieces for 3 female voices, all of which being written for the group, Scottish Voices, directed by Graham Hair. Musically, the 'subtle forces' at work are the pure sound of female voices and vibraphone, and the repeated phrases (one for each of the hexagram's 6 lines) which gradually transform, one into the next. There are certainly no 'surprise attacks', and the music is meant to conjure up a mood appropriate to its title.

*The Well*

The composition takes its title from a character (or hexagram) in the I Ching (the Chinese Book of Changes). The Well is concerned with the timelessness of existence, and is said to contain its entire history within itself. The Well is the"unchanging within change"; the constant around which all else is in a state of flux.The piece was commissioned by and composed at the University of Glasgow in 1992, and is the second in a series of three pieces for 3 female voices, all of which being written for the group, Scottish Voices, directed by Graham Hair.The tape part consists of the layering of several strands of looped patterns, which eventually form a dense and 'watery' texture. Weaving around each other, the voices rise in pitch and intensity, emphatically expressing a single idea–the meaning of which can be felt, but not conveyed in words (hence the absence of text).For the tape part, material was generated using algorithmic composition techniques, and the specific loops were arrived at after a careful process of selection and editing.The sounds were largely created on a basic FM synthesizer, the Yamaha TX-81Z.

*The Joyous*

The piece completes a series of three pieces for 3 female voices.all of which having been written for the group, Scottish Voices (directed by Graham Hair). Like its companion pieces, *The Joyous* is based on its eponymous character(hexagram) from the I Ching (the Chinese Book of Changes),the qualities of which in this case can be described as inner strength and firmness within, combined with acquiescence and softness without.The piece begins softly with a (hocket-like) pattern spread across the 3 voices, accompanied by (scale-like) figures of changing phrase lengths on the harp. The material is subjected to a variety of transformations, involving pitch, meter, rhythm,and mode, but proves ultimately to be unbreakable as its identity remains intact throughout. This musical journey (process) hopefully depicts in sound, some of the attributes embodied by The Joyous.

*Awakening*

The installation piece received its premiere at an outdoor location - the 18th-century formal gardens of Greenbank House in Glasgow, Scotland, April 1995. The 4 soundpieces (with a combined duration of over 60 mins.) were composed during a 3-month period of intensive work from January to April 1995.

*Awakening* invites the listener on a journey through sound. From a state of initial dormancy to the realisation of some ultimate goal, the four component soundpieces represent stages along the way. Non-realtime additive and FM synthesis, and effects processing of sampled sound (using the software package CLM (Common Lisp Music)), created much of the material for *Slumber*. Algorithmic methods (using the software package Common Music) in which the parametric values of events (eg. pitch, amplitude, as well as timbral details) were determined according to their position in a metrical hierarchy, generated the patterns used in *First Steps* - these triggering sounds on a Yamaha TG-77 synthesizer. *Quest* makes use of more-or-less untreated but unusual sampled sound sources (eg. recordings made inside a large empty drinking water container produced the low percussive sounds; and the rustling of a large piece of hardboard produced the percussive sound employed as a cross rhythm) and was assembled entirely using a MIDI sequencer and sampler. For *Confluence*, granulated, time-stretched, and time-compressed water samples were layered to form a slowly evolving sound texture.

**David A. Jaffe**

*The Seven Wonders of the Ancient World*

A seventy-minute concerto in seven ten-minute movements for Boie-Mathews Radio Drum-controlled Disklavier Grand piano and ensemble of plucked string and percussion instruments: mandolin, guitar, harp, harpsichord, bass, 2 percussionists, harmonium. 1-4 movements were premiered with Sonor ensemble, UC San Diego, June, 1994; 5-6 movements - University of Victoria, April 1995. 4th movement was also performed with Athelas ensemble, at ICMC 1994, Aarhus, Denmark. Work on the piece was supported in part by a Collaborative Composer Fellowship from the National Endowment for the Arts and the Banff Centre for the Arts, Canada. Recording done at the Banff Centre for the Arts, Canada 1993, 1996 will be released on commercial CD in Sept. 1996 on the Well-Tempered Productions label.

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.

**Jun Kim**

*Reverberation*

For two sopranos, percussion and computer processed sounds on tape, using CLM, SynthBuilder, SoundWorks and RT on a NeXT computer.

The piece is performed in the dark under five candlelights. The idea of this piece is based on reverbing sound effect.

**Peer Landa**

*Downcast* (1995)

For computer-generated tape. The piece was commissioned by the Norwegian Contemporary Music Organization and completed after two years of work. It was premiered in Oslo, also performed at Stanford, 1995.

The compositional technique is entirely based on digital signal processing. Several DSP applications in the C-programming language have been written exclusively for this piece, i.e. no commercial application has been used. *Downcast* serves as a presentation of these programs as well as a demonstration of a rather modern compositional technique which is a spinoff from the idea of using a general computer language (its code) as the musical notation.

The initial audio material for the piece is derived entirely from a recording of a female voice – throughout the piece, this voice is rigorously processed by the computer programs. The original sample,a recording of short laughter, can be heard in at the very end of the piece. Complex rhythmical syncopation is a crucial component for the composition. At times there are up to one thousand layers where the melody line jumps from one layer to another following the pattern of these syncopations. Elements such as dynamics and spatiality are also fundamental to the piece. Reverberant spaces are derived from actual physical rooms in the CCRMA-building (everything from the smallest closet to a large auditorium have been used for reverb impulse-responses). The convolution of those room-responses are combined into layers and used in the style of classical counterpoint. Since the composition is entirely processed and edited in the digital domain (no analog to digital converters have been used) the sound is significantly clean with a very high dynamic range.

**Lukas Ligeti**

*New Music for Electronic Percussion* (1996)

**Fernando Lopez Lezcano**

*Three Dreams*

a) *Paper Castles*

b) *Invisible Clouds*

c) *Electric Eyes*

This piece is about impossible dreams. Many times and without learning from experience we build beautiful Paper Castles on Invisible Clouds, thinking yet again that dreams are reality, or maybe that they can be turned into reality with sheer will power and a magical wand. This sections are like twin brothers, intermingled yet separate. As for the last section, Electric Eyes, if one has ever felt the startling contact of electric eyes, there is no need for the composer to explain. If one has not, mere words will never be enough. That's the composer's dream and the cause of a lot of paper castle building...

The piece was composed in the digital domain using the CLM non real time sound synthesis and processing environment running on a NeXT and the four channel spatialization was performed by a special unit generator programmed by the composer. The original sound materials are sampled tubular bells, cowbells, cymbals, gongs, knives and screams and quite simple additive synthesis instruments. The first part was composed while the author was working in Japan at the Computer Music Laboratory of Keio University. It was latter finished at CCRMA.

*Espresso Machine II*

*Espresso Machine II* is the second incarnation of the first piece that uses PadMaster, a new improvisation / performance environment built around the Mathew/Boie Radio Drum and written by the composer on a workstation running the NextStep operating system and a live electronic cello player (Chris Chafe playing his electronic Celletto). PadMaster is written in Objective-C and uses the MusicKit classes as the basic foundation for MIDI control and sequence playback. The Radio Drum interfaces with the NeXT through a custom MIDI protocol and is used to trigger and control isolated events and event sequences in real time. PadMaster splits the drum surface into programmable virtual pads that can be grouped in sets or "scenes", which in turn represent different behavioral patterns for the different sections of the piece.

*Espresso Machine* is an evolving dialog between the acoustic / electronic sounds of the Celletto and the contrasting timbres played by the composer on two TG77 synthesizer modules through the PadMaster program controlled by the Radio Drum. PadMaster essentially provides several pallettes of pre-built elements that are combined and controlled in real time to generated an electronic soundscape for the Celletto performance.

*Knock Knock... anybody there?*

"Knock Knock... anybody there?" is an extension to four channels of the original stereo sound track that was composed for a collaboration project with visual artists in 1994. Willie Scholten and Ruth Eckland provided the sculptures and visual framework while this piece served as the sound environment for the installation. The music explores altered states of consciousness and in particular insanity, in a journey through a three dimensional soundscape where voices and sounds evoke multiple and conflicting states of mind. All the concrete sound materials used in the piece were gathered during a small meeting with friends where the central topic that motivated the project was freely discussed. From the digital recording small but significant fragments of the conversation were extracted and subsequently processed in the digital domain using CLM instruments (CLM, Common Lisp Music, a non real-time Lisp based software synthesis and processing environment). The processing included dynamic spacialization of multiple moving sources rendered for a four channel reproduction environment. The listener moves through the soundscape while voices and sounds tell several overlapping stories that might occur in the hazy border between sanity and insanity. The piece even includes materials from the piano jam session that happened at the end of the meeting...

*With Room to Grow*

"in a room - with room to grow - the fabric of space is floating veils, curtains and webs... alabaster light and tides of time play with them. Grandma is sitting in a rocking chair... she looks at me, smiles, and keeps knitting an infinite tapestry of gifts"

This is a solo piece for PadMaster (a real-time improvisation software package written by the composer), Radio Drum and MIDI synthesizers. PadMaster uses the Mathews/Boie Radio Drum as a three-dimensional MIDI controller. The function of the batons and the behavior of the surface of the drum are controlled through PadMaster and create a set of sonic soundscapes through which the performer chooses a path.

**Jose Montalvo**

Compositions completed at CCRMA: *Mito Caribe* for large symphonic orchestra; *Sea Shells* for Perry Cook's collection of shells; *Totems* for multi media; *2x1* for solo percussionist with vibraphone and marimba. He is currently working on a piece for string orchestra.

**Jonathan Norton**

*Vicissitudes* (1995)

For computer-generated stereo tape.

**Juan Carlos Pampin**

*Apocalypse was postponed due to lack of interest* (1994)

The piece was composed at CCRMA, Stanford University, during the summer of 1994. The generation, transformation, and mixing of the sounds for the compostion were done in a NeXT computer using Bill Schottstaedt's CLM. Its stucture presents a continuous evolution of a group of materials. Sounds objects undergo different kinds of mutations in short and long term, creating by their interaction textures with distinct morphology.

Performed at the following locations: Semana de la Musica Electroacustica, October 1994, Buenos Aires, Argentina; Stanford University, November 1994; Vibrations Composees, April 1995, Lyon, France; International Computer Music Festival, May 1995, San Diego; Synthese: 25e Festival de Musique Electroacoustique, June 1995, Bourges, France; Punto de Encuentro IV Festival Internacional de Musica Electroacustica, December 1995, Madrid, Spain; Universidad Catolica Argentina, December 1995, Buenos Aires, Argentina. The composition received an award at the 22e Concours International de Musique Electroacoustique, Bourges, France, 1995.

*Transcription No.1* (1996)

For computer-controlled Disklavier. The textures and rhythms for this piece were generated with Common Music, using a granular/additive synthesis algorithm. The spectrum (harmony) and formants (dynamics) for the piece were derived from an analyzed sound, using the composer's own ATS (Analysis/Transformation/Synthesis) software. A vocal tone, transformed by means of transposition and frequency shifting, was used as a formal metaphor for the whole piece.

**Fiammetta Pasi**

*Collage* (1995)

For computer-generated stereo tape. As the word 'collage' suggests, similarly to the work in the visual arts which is made by putting together various 'patches of color', this short piece is based on creating and overlapping many 'patches of sounds'. This work is the result of explorations in several different computer environments including CSOUND, STELLA and Music Kit, CLM where the basic materials, the timbres ("instruments"), are realized through additive and FM synthesis. The piece has been composed not accordingly to a pre-established project, but proceeding with little sections, fragment by fragment, leaving any possibility open, and with the constant intention to always keep the internal movement and energy alive.

**Jorge Sad**

*Vox* and *VoxII* (1995)

Both computer-generated compositions were premiered at Stanford University in 1995 and later the same year presented at Centro Cultural Recoleta, Buenos Aires. *VoxII* was also performed at International Days of Electroacoustic Music, Cordoba, Argentina, December, 1995 and received Juan Carlos Paz electroacoustic music prize 1995 granted by the Fondo Nacional de las Artes, Argentina.

The basic aim while composing these pieces was continuing the exploration of sound materials of ethnic and art rock music. As copyright laws are not concerned with large musical structures (forms) or very small ones (sounds), Jorge Sad worked in the border zone in which a sound or small group of sounds are still

recognizable as belonging to a particular style, composer or player but is integrated in a totally different musical context.

### Bernd Hannes Sollfelner

*"Das Tor zur Hoelle/Plastophonic World. Hommage a A.Rodin* (1994-1995)

A computer-generated tape piece, premiered at Stanford University, April 16, 1995, performed in Vienna, Austria, 1995-1996.

### Marco Trevisani

*Up to four and more* (1994)

Nine compositions for prepared piano, drum, real time sound processing, computer generated sounds on tape. Performed by Marco Trevisani, Lukas Ligeti, Nick Porcaro, Michael Edwards.

*'O core dumped* (1995)

For prepared piano, prepared guitar and computer generated sounds on tape . Performed by Marco Trevisani, Davide Rocchesso and Michael Edwards.

*Cosi' sia* (1995)

A computer theatre/music production for actors, singer, live electronic and computer triggered sounds/mixing with ArtiMix. Based on a Pirandello's play.

*Frammenti e Variazioni su Aura* (1995)

A computer generated tape composition, inspired by Bruno Maderna's Aura.

*Segmentation Fault beta1.0* (1996)

For prepared piano and computer triggered sounds/mixing with ArtiMix. Performed by Marco Trevisani and Michael Edwards.

All pieces except *Cosi' sia* have been performed at CCRMA, Stanford University concerts. *'O core dumped* was also performed in Padova, Italy and at the Kitchen, New York. *Cosi' sia* will be presented in winter 1996 in Italy.

# Chapter 6

# Research Activities

## 6.1 Overview

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.

## 6.2 Computer Music Hardware and Software

### 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 behaviour.

## 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.

## ATS: Analysis/Transformation/Synthesis; A Lisp interface for SMS (Xavier Serra's Spectral Modeling Synthesis) and CLM (Schottstaedt's Common Lisp Music)

Juan Carlos Pampin

ATS intends to be a sound composition/design environment. It consists of a set of Lisp functions, allowing transformation of sound spectral information and resynthesis. Sounds in ATS are sinusoidally modeled, using SMS for the analysis part. The synthesis part is based on two instruments written in CLM: one supporting additive synthesis and the other subtractive synthesis. Both instruments use the inverse FFT technique.

Part of the work in this project was done at the SONVS center of the Conservatoire Superieur de Musique de Lyon, France, as a thesis work for a Master in Computer Music.

experimentation and repeatable results are possible. The vocal tract filter is modeled by a waveguide filter network. Glottal pulses are stored and retrieved from multiple wavetables. To this periodic glottal source is added a filtered pulsed noise component, simulating the turbulence which is generated as air flows through the oscillating vocal folds. To simulate the turbulences of fricatives and other consonants, a filtered noise source can be made arbitrarily resonant at two frequencies, and can be placed at any point within the vocal tract. In the real-time DSP program, called SPASM, all parameters are graphically displayed and can be manipulated by using a computer mouse. Various two-dimensional maps relating vowels and vocal tract shapes are provided, and a user can smoothly vary the control parameters by moving the mouse within a map region. Additional controls include arbitrary mapping of MIDI (Musical Instrument Digital Interface) controls onto the voice instrument parameters. The software synthesis system takes as input a text file which specifies the events which are to be synthesized. An event specification includes a transition time, shape and glottal files as written out by the SPASM system, noise and glottal volumes, glottal frequency (either in Hz or as a musical note name), and vibrato amount. Other control strategies available include text-to-speech/singing and a graphical common music notation program. Support for languages, musical modes, and vocal ornamentations is provided in Latin, and modern Greek.

## 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.

## The "Flutar" a New Instrument for Live Performance

Cem Duruoz

"Flutar" is a cross-synthesis instrument which consists of a physical simulation of the flute* combined with a live instrument, in particular the classical guitar. It is implemented by using the software "SynthBuilder" on a Next computer. During a live performance, a second computer modifies its parameters in real-time, or in other words "plays" the "flutar", while the performer plays the guitar. The physical model for the simulation combines an excitation section and a resonator, which correspond to the embouchure and the bore of a real flute, respectively. The two instruments interact with each other during the performance. In other words, the sound that the computer generates is dependent on the guitar sound that it receives by means of a microphone: the amplitude of the guitar sound modifies the input noise that simulates the wind blowing into a flute. At the same time the captured guitar sound goes through the resonator to produce the impression of a "plucked flute". This way, there may be resonances which emphasize the guitar sound depending on the pitches played by the guitar as well as the pitch that the flutar is tuned to.

## Spectral Operators for Timbral Design

Jose Eduardo Fornari

The research that have been developed in this doctoral course of Electrical Engineering Faculty of University of Campinas, FEE/UNICAMP, to conclude the thesis entitled: "Spectral Operators for Timbral Design" follow the previous master thesis research of the same author whose title is"Sound Transformations by Timbral Operations". In this master thesis, it was developed a DSP system that works in real time to synthesize sound timbres by parametric transformations on the sound frequency spectrum. These transformations were called like "Timbral Operations", TO.

The current work develops the TO original concept from the creation and cataloging of families of basic spectral sound operations, called SOs, with whom it's supposed to create a methodological organization for TOs used to timbral projects. This methodology will be organized in one language for timbral modeling, or design, so called SOFTD (Spectral Operators for Timbral Design).

Through the SOFTD it will be possible to carry out timbral modelings in on-line, to anyone sound. This modeling occurs in real time and preserve the others sound features (as loudness and pitch) unchanging. Timbre is therefore the only sound feature manipulated by SOFTD.

SOFTD will be based on algorithmic structuring of SOs, where each algorithm represents one TO. The SOs will represents functions whose both parametrization and action will be in real time. The algorithms (TOs) shall be modified in on-line by the system user. Therefore, SOFTD will allow the intuitive learning of timbral modeling.

SOFTD will be incorporated with the previous project made in the master thesis research, or else, in one real time DSP system whose input is a generic sound and output is the same sound but with its timbre modified. Through the hardware project of SOs, this system will have real time action. Then SOFTD will be a software language of algorithmic structuring of these SOs where will be hardware implemented to allow real time timbral modelings.

## Voice Gender Transformation with a Modified Vocoder

Yoon Kim

The goal of this project is to develop a voice transformation system that makes the transformed voice close to a natural voice of the opposite sex. The transformation considers the differences of fundamental frequency (pitch) contours and spectral characteristics.

The transformation algorithm employs components of a vocoder well known as the LPC-10 vocoder. By using the analyzer and the synthesizer of the LPC-10 vocoder and inserting the transformer in between them, we can modify the LPC analysis parameters at the transformer stage, and so change the acoustic nature of the input speech by feeding the modified parameters into the synthesizer.

In converting the gender of a voice, two parameters – pitch and formants– are modified. Pitch is transformed by viewing the pitch as a random variable, and changing the mean and standard deviation of the original pitch values. Formant frequency is defined as the frequency corresponding to a peak of the speech spectrum, whlie formant bandwidth is defined as the 3-dB bandwidth of the peak. The first three formant frequencies

are scaled separately by empirically derived factors. The scale factors for formant bandwidths are set equal to those for formant frequencies.

Based on the above ideas, an algorithm for voice gender transformation is implemented. Its performance depends greatly on the original speaker. Also, female-to-male conversion was found to produce more natural sounding speech compared to male-to-female conversion. This is mainly due to the fact that the LPC-10 vocoder is poor in synthesizing female voice.

## Processing of Critically Sampled Audio Subband Data

Scott Levine

Digital audio data is being transmitted over more types of media and to more people than ever before in such diverse areas as satellite television, digital radio, cable, movies, and via the internet. Since bandwidth is always at a premium, the digital audio is data compressed by factors ranging from 4:1 to 20:1, depending on the application. These compression algorithms usually have very complex, expensive encoders, while the decoders are simple and inexpensive. The commonly used encoders, which today has boiled down to MPEG and AC3, transmit the audio decomposed into many separately quantized, critically sampled, bandpassed subband signals. In addition, they transmit some side information about the signal, derived from some encoder's psychoacoustic model, such as subband bit allocation and transient detection.

At the decoder, which can now run realtime on today's Pentium PCs, the original audio is approximately reconstructed from the quantized subbands and the side information. But after decoding, the audio is sometimes run through effects processing algorithms such as reverberation, chorusing, vocoding, or time stretching. It has been found that these post-processing algorithms can be commuted back within an MPEG audio decoder. These algorithms are slightly modified to run on the individual quantized subbands. One of the benefits to this approach is that since the data rate has been reduced, the amount of memory and computations required for the effects is equally reduced. Another advantage is that the effects designer can tailor the algorithm to behave differently in separate subbands, since the compressed audio is transmitted in bandpassed subband signals. These post-processing algorithms can also utilize the side information that a complex audio encoder calculated somewhere else. For example, the effects' memory and computation allocation can be dynamically scaled as a function of the psychoacoustic bit allocation and the transient detection.

# Feedback Delay Networks

Davide Rocchesso

Recursive comb filters are widely used in signal processing, particularly in audio applications such as digital reverberation and sound synthesis. In the recent past, some authors [Stautner-Puckette '82, Jot '92] have considered a generalization of the comb filter known as the feedback delay network (FDN). The main purpose of this research is to investigate the algebraic properties of FDNs as well as to propose some efficient implementations and interesting applications.

The FDN is built using N delay lines, connected in a feedback loop through a set of scattering coefficients. These coefficients may be organized into a "feedback matrix". If such a matrix is unitary, system poles have magnitude one and the MFDN has only constant-amplitude eigenmodes. For the structure to be practically useful, an attenuation coefficient must be applied at the output of each delay line to adjust the length of the impulse response.

D. Rocchesso has proposed restricting the feedback matrix to a circulant structure. The resulting Circulant Feedback Delay Network (CFDN) can be efficiently implemented and allow an easy control of the time and frequency behavior. This structure is also proper for VLSI implementation because it can be efficiently made parallel.

It has been shown how to use CFDNs for many purposes in sound processing and synthesis: for simulation of radiating structures such as instrument bodies, for simulation of feedback resonators, and even for live electronics performances. These possibilities extend the range of applicability of FDNs beyond reverberation.

CFDNs with short delay lines may be used to produce resonances irregularly distributed over frequency. A possible application could be the simulation of resonances in the body of a violin. In this application the exact position and height of resonances are not important. By changing delay lengths, it is possible to move poles in frequency, while by changing the network coefficients we can re-shape the frequency response. The loop gain determines the maximum peak to valley distance. Such a structure using short delay lines has been used in live-electronic-sound processing, where a dynamic filtering can be achieved by changing the FDN parameters in real time.

CFDNs are also very effective as resonators in Karplus-Strong-like algorithms, especially for simulating membranes or bars.

Connections between FDNs and Digital Waveguide Networks (Smith '85) have been revealed. Julius O. Smith and D. Rocchesso have shown that the FDN is isomorphic to a (normalized) waveguide network consisting of one (parallel) scattering junction and N branches, each connecting to the one scattering junction at one end, and reflectively terminated at the other. This correspondence gives rise to new generalizations in both cases. Theoretical developments in this field are in progress.

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## Acoustical Research on Reed Driven Woodwind Instruments for the Purpose of Efficient Synthesis Models

Gary 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 platform for acoustical explorations and research. Realistic clarinet models have been developed (Cook), as well as an interactive development environment (Hirschman) for the study of reed dynamics. To date, the clarinet has drawn most of the attention, mainly because of its relative simplicity. Models of the saxophone have been scarce, though the recently released Yamaha VL1 demonstrates that the saxophone too can be effectively modeled using waveguide techniques.

Simple woodwind models have been designed and implemented in the NEXTSTEP programming environment and also using SynthBuilder. Work in progress includes the addition of more advanced expressive controls to current models, as well as implementation of toneholes. Further, the effect of oral cavity manipulation in the performance of woodwind instruments is being studied. Results of this study will be incorporated into woodwind synthesis models.

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# 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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# 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.

A basic feature of DWF building blocks is the exact physical interpretation of the contained digital signals as samples of traveling pressure waves, velocity waves, or the like. A by-product of this formulation is the availability of signal power defined instantaneously with respect to both space and time. 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. Another nice property of waveguide filters is that they can be reduced in special cases to standard lattice/ladder digital filters which have been extensively developed in recent years. One immediate benefit of this connection is a body of techniques for realizing 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. Waveguide filters can be viewed as providing a discrete-time "building material" incorporating aspects of lattice and ladder digital filters, wave digital filters, one-dimensional waveguide acoustics, and classical network theory. Using a "mesh" of one-dimensional waveguides, modeling can be carried out in two and higher dimensions. (See Van Duyne, Smith, p. 23.)

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. 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. 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.

When the wave impedance changes, signal scattering occurs, i.e., a traveling wave impinging on an impedance discontinuity will partially reflect and partially transmit at the junction in an energy-preserving way. Real-world examples of waveguides include the bore of a clarinet, a violin string, horns, organ pipes, the vocal tract in speech, microwave antennas, electric transmission lines, and optical fibers.

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## A Passive Nonlinear Filter for Physical Models

John Pierce and Scott Van Duyne

Nonlinearities, small or large, favorably affect the sounds of many musical instruments. In gongs and cymbals, a gradual welling-up of energy into the high frequencies has been observed. Nonlinearities cause the transfer of energy from lower modes to higher modes after the instrument has been struck. These nonlinearities do not generate new energy, only transfer it. While memoryless square–law and look–up table nonlinearities may be incorporated in computer generation of sounds, these means often cause system energy loss or gain, and are difficult to control when a range of large and small effects are desired.

Our approach to the injection of nonlinearity into resonant systems was to identify a simple passive nonlinear electrical circuit, and then to apply physical modeling techniques to bring it into the digital signal processing domain. The result was an efficient digital nonlinear mode coupler which can be attached to any waveguide termination, or inserted into any resonant digital system where traveling waves are being computed. The mode coupler can be tuned to set the rate of energy spreading as well as the region of the spectrum to be affected. Excellent results have been obtained creating gong and cymbal crash sounds by connecting these passive nonlinear filters to 2–D Digital Waveguide Mesh boundary terminations.

This work has been presented by Scott Van Duyne at the 1994 and 1995 ICMC and at the Washington D.C. meeting of the Acoustical Society of America. 30 May - 3 June. 1995.

Related matters remain under investigation.

## The Digital Waveguide Mesh

Scott Van Duyne and Julius Smith

The traveling wave solution to the wave equation for an ideal string or acoustical tube has been modeled efficiently with bi-directional delay-line waveguides. Two arbitrary traveling waves propagate independently in their respective left and right directions, while the actual pressure at any point may be obtained by summing the theoretical pressures in the left– and right–going waves.

Excellent results have been obtained modeling strings and acoustic tubes using one-dimensional waveguide resonant filter structures. However, there is a large class of musical instruments and reverberant structures which cannot be reduced to a one-dimensional traveling wave model: drums, plates, gongs, cymbals, wood blocks, sound boards, boxes, rooms—in general, percussion instruments and reverberant solids and spaces.

In the two dimensional case of wave propagation in an ideal membrane, the traveling wave solution involves the integral sum of an infinite number of arbitrary plane waves traveling in all directions. Therefore we cannot just allocate a delay line for every traveling plane wave. Finite element and difference equation methods are known which can help with the numerical solution to this problem; however, these methods have had two drawbacks: (1) their heavy computational time is orders of magnitude beyond reach of real time, and (2) traditional problem formulations fit only awkwardly into the physical model arena of linear systems, filters, and network interactions.

Our solution is a formulation of the N-dimensional wave equation in terms of a network of bi-directional delay elements and multi-port scattering junctions. The essential structure of the two-dimensional case is a layer of parallel vertical waveguides superimposed on a layer of parallel horizontal waveguides intersecting

each other at 4-port scattering junctions between each bi-directional delay unit. The 4-port junctions may be implemented with no multiplies in the equal impedance case. Plane waves, circular waves, and elliptical waves all propagate as desired in the waveguide mesh. Band limited accuracy can be enforced. The three-dimensional extension of the waveguide mesh is obtained by layering two-dimensional meshes and making all the 4-port junctions into 6-ports, or though a tetrahedral, four-port, no-multiply structure.

The two-dimensional waveguide mesh is mathematically equivalent to the standard second-order-accurate finite partial difference formulation of the wave equation. It, therefore, exhibits the desirable stability and convergence properties of that formulation. However, the numerical solution methods of initial value problems involving second order hyperbolic partial difference equations usually require a multi-step time scheme which retains values for at least two previous time frames. The waveguide mesh reduces this structure to a one-step time scheme with two passes: (1) In the computation pass, the scattering junction computations are performed in any order (a feature well–suited to parallel computation architectures); then (2) in a delay pass, their outputs are moved to the inputs of adjacent junctions.

Current work on the waveguide mesh is in (1) exploring alternative spatial sampling methods, (2) developing efficient hardware implementation structures, (3) introducing loss and dispersion into the mesh in a physically correct, yet efficient, manner, and (4) finding the right parameters to model specific musical instruments and spaces.

## The Wave Digital Hammer

Scott Van Duyne and Julius Smith

Recent work has led to digital waveguide string models and physical models of membranes using a 2–D digital waveguide mesh. We are currently working on ways to excite these models in physically correct ways. One obvious need is a good model of the felt mallet for drums and gongs, and of the piano hammer for strings.

The attack transient of a struck string or membrane can be approximated by the injection of an appropriate excitation signal into the resonant system. However, this excitation method is not sufficient to cope with the complexities of certain real musical situations. When a mallet strikes an ideal membrane or string, it sinks down into it, feeling a pure resistive impedance. In the membrane case, the depression induces a circular traveling wave outward. If the membrane were infinite, the waves would never return, and the mallet would come to rest, losing all its energy into the membrane. If the membrane is bounded, however, reflected waves return to the strike point to throw the mallet away from the membrane. The first reflected wave to reach the mallet may not be sufficiently powerful to throw the mallet all the way clear, or may only slow down its motion, and later reflected waves may finally provide the energy to finish the job. This complex mallet-membrane interaction can have very different and difficult to predict acoustical effects, particularly when a second or third strike occurs while the membrane is still in motion.

In our model, we view the felt mallet as a nonlinear mass/spring system, the spring representing the felt portion. Since the felt is very compliant when the mallet is just barely touching the membrane, yet very stiff when fully compressed, we must use a nonlinear spring in the model, whose stiffness constant varies with its compression. Our essential requirements are that the model be digitally efficient, that it be easily interconnected to waveguide structures, and that it be able to compute arbitrarily accurate strike transients from measured data from real hammers, strings, mallets, and drums.

# The Commuted Waveguide Piano

Scott Van Duyne and Julius Smith

Making a good piano synthesis algorithm traditionally has not been easy. The best results thus far have been in the area of direct sampling of piano tones. This approach is memory intensive, as there is a lot of variety in the piano timbre ranging from low notes to high, and from soft to loud. In addition, sampling techniques don't have a good answer to the problem of multiple strikes of the same string while it is still sounding, nor to the coupling of strings which are undamped while other strings are sounding. We believe that the solution will be found through waveguide modeling and DSP synthesis techniques. An even more intrinsic problem of synthesizers is that they don't feel anything like a piano when you play them. Currently at CCRMA, we have the good fortune of having a variety of people working separately on solutions to different parts of the piano problem, although their individual work may have broader applications.

The piano problem may be broken down into five basic parts: (1) the string, (2) the soundboard, (3) the piano hammer and damper, (4) the key mechanism itself, and (5) the implementation hardware and software.

The primary difficulties of modeling the string are found in that the piano string harmonics are not exactly harmonic, and in that there is significant coupling between horizontal, vertical and longitudinal modes on the string. In addition, there may be important nonlinear effects. Work being done by Julius Smith on string coupling and fitting loop filters to measured data will solve some of the problems. Other work by Scott Van Duyne and Julius Smith will lead to simplifications in modeling the stretching harmonics of the piano string. It may be that work relating to passive nonlinearities by John Pierce and Scott Van Duyne will be helpful for the finest quality tone generation.

The soundboard can now be modeled in a fully physical way using the 2-D Digital Waveguide Mesh, a recent development by Scott Van Duyne and Julius Smith which extends waveguide modeling techniques into two or more dimensions. Julius Smith is working on applying results from his work on bowed strings to piano synthesis; extremely efficient new algorithms are possible using this approach.

The excitation of waveguide string models has till now been left primarily to loading the string with an initial condition and letting it go, or to driving the waveguide loop with an excitation signal tailored to the desired spectral response of the attack transient. While almost any attack transient may be achieved through driving the model with an excitation signal, the variety of interactions that a piano hammer may have with a string is immense when one considers the possibilities ranging over the very different high and low strings, and over the wide range of strike forces. Further, it would be virtually impossible to catalog the possible attack transients due to a hammer hitting a string which is already in motion due to a previous strike. The hammer/string interaction is very complex. Fortunately, recent work initiated by Scott Van Duyne and continued with Julius Smith on modeling piano hammers as wave digital nonlinear mass/spring systems will allow all these complex interactions to fall directly out of the model.

Brent Gillespie's work on the touchback keyboard project will provide a realistic controlling mechanism for the piano synthesis algorithm. The touchback keyboard is a haptic control mechanism, driven by a computer controlled motor. It looks like a piano key, and feels like a piano key. That is, it senses the force applied by a person to the key, and computes, in real time, the correct key motion response based on the equations of motion of the internal key mechanism. It is easy for the touchback keyboard to provide the felt hammer element of the tone synthesis algorithm with a hammer strike velocity. This velocity will be used to drive the synthesis algorithm. In return, the piano hammer element can provide the touchback keyboard with a return hammer velocity at the right time, and the person playing the key will feel the appropriate haptic response.

The hardware and software to implement this complete piano model is available now. The touchback keyboard is controlled by a PC with an add-on card dedicated to real-time computations of the equations

of motion. The NextStep operating system running on a Next or PC platform will provide a suitable environment for the synthesis algorithm. Specifically, the SynthBuilder Application being developed by Nick Porcaro and Julius Smith provides a cut-and-paste prototyping environment for real-time DSP-based audio synthesis, and the Music Kit, being maintained and improved by David Jaffe, provides higher-level access to the DSP 56000 card.

There is additional research interest in vibrotactile feedback in the piano keys as suggested in the current work of Chris Chafe. While this effect may be less important in the modern piano, it is certainly more important in early keyboard instruments, and critical in the clavichord, where the hammer may remain in contact with the vibrating string after striking it. Further, we shall want to make the piano sound as if it were somewhere in a particular room or concert hall. Work by John Chowning in localization, by Jan Chomyszyn in loudness perception, by Steven Trautmann in speaker arrays, and by R. J. Fleck in efficient reverberation models can round off the final auditory experience.

## 6.4 Controllers for Computers and Musical Instruments

### Real-time Controllers for Physical Models

Chris Chafe and Perry Cook

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

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

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

New pieces are being written using real-time controllers and the DSP-based physical models. "El Zorro" is a recent composition by Chris Chafe that employs a Lightning Controller (by Buchla and 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.

### Ongoing Work in Brass Instrument Synthesizer Controllers

Perry Cook and Dexter Morrill

Brass instrument players have been at a disadvantage when using their instruments as computer music controllers, because they have been limited to commercial pitch extractors which do not measure and use the unique spectral and control features of the brass instrument family. In this project, brass instruments were fitted with several sensors and were used in conjunction with specially designed pitch detection algorithms.

Systems were constructed using various valved brass instruments. Pressure sensors located in the mouthpiece, on the bell, and in mutes are used for pitch detection and pickup of the direct horn sound. Switches and linear potentiometers were mounted near the valves for finger control, and traditional foot pedals are also available for control. Optical sensors were mounted on the valves, providing information about valve position.

The valve and acoustic information are used to form pitch estimates which are both faster and more accurate than those yielded by commercial pitch extractors. The other switches and controls provide control over MIDI synthesizer parameters such as sustain, patch change, and controller changes, as well as controlling signal processor parameters.

## The Touchback Keyboard

Brent Gillespie

A virtual piano action (a mathematical model), a simulation algorithm, and a set of 88 single degree-of-freedom haptic display devices constitute a promising means to make a synthesizer keyboard feel like a real grand piano. Other keyboard instruments could be simulated with the same device at the touch of a button. We propose, then, to simulate the feel of the grand piano by numerically integrating the equations of motion of the multibody piano action in real-time in a human-in-the-loop simulation scheme. As the integration proceeds, the finger-key interaction force is computed and generated by a haptic interface. In addition, the motion (needed for use in the integration) is sampled from the haptic interface hardware.

We believe that haptic (tactile/kinesthetic) feedback from an instrument, in addition to the aural feedback, plays a crucial role in the process by which a keyboardist controls the tone and timing. Further experiments in human-computer interaction can be made with arbitrarily determined dynamical behaviors of the keyboard.

The behavior of the simulated piano action should appear passive to the user even though it is realized with active devices. We seek design guidelines which will ensure that such dynamical behaviors characteristic of active objects (chatter and stickiness) will not arise. Applications of both linear and non-linear analytical tools are being applied to this end.

## The Computer-Extended Ensemble

David Jaffe

Until recently, there have been two basic models of how electronics interact with a performer in a performance situation. One model adds a tape of synthesized sound to an instrumental ensemble. We call this the "tape music" model. The other model, "keyboard electronic music," consists of pianists performing on keyboard synthesizers. In the case of the tape music model, the performer is forced to slave to the electronics; with keyboard music, it is the electronics that slave to the performer. We are beginning to realize that these two models are actually end points of a continuum, with the region between them largely unexplored.

The central question for composers is not whether human behavior can be duplicated, but what new musical effect can be achieved with computer interaction that cannot be achieved by prior existing means. A likely place to begin exploring this question is in an area of music in which interaction between performers is central – improvisation.

Introducing a computer as an extension of the improvising performer increases the scope of spontaneous musical decision-making that gives improvisational music its distinctive quality. A computer can magnify, transform, invert, contradict, elaborate, comment, imitate or distort the performer's gestures. It gives the performer added power to control sound at any scale, from the finest level of audio detail to the largest level of formal organization.

But the full power of the computer in an improvisational context does not show itself until we add a second performer to the ensemble. Now each performer can affect the playing of the other. One performer can act as a conductor while the other acts as soloist. Both performers can be performing the same electronic instrument voice at the same time. And these roles can switch at a note-by-note rate. Thus, the walls that normally separate performers in a conventional instrumental ensemble become, instead, permeable membranes. Figuratively speaking, the clarinetist can finger the violin and blow the clarinet while the violinist bows the violin and fingers the clarinet. We have coined the term "computer-extended ensemble" for this situation.

The challenge becomes finding roles for the performers that allow them just the right kind of control. They need to feel that they are affecting the music in a significant and clear fashion. Otherwise, they will feel superfluous and irrelevant, as if the music has gotten out of control. The computer program may be simple or complex, as long as it fires the imagination of the performers.

We have been experimenting in this realm with percussionist/composer Andrew Schloss in a duo called *Wildlife*. The duo features Schloss and the author performing on two modern instruments, the Mathews/Boie Radio Drum and the Zeta electronic/MIDI violin, with this ensemble extended by two computers, a NeXT and a Macintosh running the NeXT Music Kit and Max. The music is a structured improvisation in which all material is generated in response to the performers' actions; there are no pre-recorded sequences or tapes.

Current work includes *The Seven Wonders of the World*, which, unlike *Wildlife*, casts the computer and Radio Drum in the context of a conventional ensemble (or, at least, a conventionally notated ensemble). This piece is scored for Radio Drum-controlled Disklavier, harpsichord, harp, two percussionists, mandolin, guitar, harmonium and bass. It was composed at the Banff Centre for the Arts, where I was a Visiting Artist in 1992-93. The Radio Drum part was worked out in collaboration with Andrew Schloss, supported by a Collaborative Composer Fellowship from the N.E.A.

Other projects include the following:

1. *Terra Non Firma*, a work for conducted electronic orchestra and four cellos, using the Mathews Conductor program. This work was commissioned by the University of Victoria in honor of Max Mathews.

2. *American Miniatures*, a recently completed work for tape alone, uses Common Music, the Music Kit and the phase vocoder to process recorded sounds of strings, voices and drums.

## Haptic User Interfaces for the Blind

Sile O'Modhrain and Brent Gillespie

Advances in graphic output technology have opened the window to 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 a simple powered joystick built from spare parts we have produced such effects as a virtual button and a variety of texture cues. Those who have used this haptic prototype agree that we have begun to tap a very promising resource.

We aim to develop an application for the PC which will be capable of making the graphic output of commercial sound editing programs accessible by touch. To this end, we are writing routines to read and recognize icons from video memory. We can already read arbitrary buttons from the screen and display them through sound (midi) and touch (powered joystick).

## The Radio Baton Progress Report

Max Mathews

### MIDI Hardware

During the last year, a new design has been completed which involves a flexible antenna, and a separate electronics box with a LCD display.

Previous hardware had the receiving antennas located under the upper surface of a large "pizza" box which also contained the electronics. Ergonomic considerations required that this box be about 2 feet square–a size large enough to be inconvenient to pack and transport and expensive to manufacture. In the new hardware the electronics are in a much smaller box, about 10 inches square and the antenna is connected to the electronics via a unpluggable cable. In addition, the antenna is flexible and can be rolled up for storage and shipping.

The breakthrough which made a flexible antenna possible was the use of aluminized mylar for the electrodes. This material is inexpensive, available, and tough. The electrodes are fabricated from a single sheet which comes completely coated with aluminum. Insulating channels are cut through the aluminum with a hand grinder. As we have already said, baton tracking is done in a very simple and robust way. A transmitter antenna electrode is located in the end of each baton. The strengths of the signals received at the five electrodes are compared.

Wires are connected to the electrodes by first attaching to the aluminum layer, small pieces of conductive-adhesive backed copper tape manufactured by the 3 M company. Wires can then be soldered to the copper. The conductive adhesive backed copper tape has proven to be a very useful material for many purposes.

In the finished receiver, two sheets of mylar, one for the electrodes, the other for a ground plane shield are between two sheets of vinyl covered fabric and the layers are sewed together around the outside edges with a sewing machine. This fabrication is easy and can be done by any clothing manufacturer or sailmaker.

The electronics box contains the 80C186 embedded processor which is the main computer in the baton. The knobs and push buttons which were part of the original baton design are also on the electronics box. A LCD display has been added to provide feedback to the performer for various purposes.

### Type 1 MIDI Files

Communication with the Radio-Baton is done entirely with MIDI characters. In the conductor program mode, the score of the piece to be played can be loaded into the Radio-Baton from a control-computer encoded as a sequence of MIDI characters.

Last year I described a version of the program that could play MIDI files. This ability has become very important. Midi files can be prepared by many commercial sequencer programs and the files appear likely to become the "lingua franca" of music representation. Both type 0 and type 1 MIDI files exist. Last year's

conductor program could play only type 0. In a type 0 file, all the events in all the channels are sorted into proper time sequence before being recorded on the file. Thus a type 0 file is simple to play with device like the Radio-Baton. Unfortunately not many sequencer programs will write type 0 files, and those that do appear to have serious bugs.

Consequently I revised the conductor-program so it can play type 1 files. The revision is non-trivial. Type 1 MIDI files have a plurality of tracks. In each track, events are sorted into their proper time order, but in order to play the complete piece, all tracks must be melded together with all events in one single proper time order sequence. A sorting program to do this task was written. The resulting program tried on a number of commercial MIDI files seems unproblematic.

### Controller Mode

The Jaffe-Schloss program allows the Radio-Baton to be used as a simple controller. In this mode, the baton sends triggers and information about the motions of the batons to a control computer. The musical interpretation of the baton signals is done in the control computer which then sends MIDI commands to a synthesizer to play the music.

The trigger and position information must be encoded as standard MIDI commands because all communication with the baton is via MIDI. The original Jaffe-Schloss protocols were somewhat prodigal in their use of MIDI commands and channels. The Jaffe-Schloss protocols have been revised. All communication is done by aftertouch and control change commands sent on MIDI channel 16. This choice was made as one least likely to preempt normal MIDI needs for these commands.

### Environment for Compositional Algorithms in the C Language

The Jaffe-Schloss mode was originally made to interface with the MAX language running in a Macintosh computer. During the winter quarter, Chris Chafe and I gave a course focused on writing compositional algorithms in the C language. To simplify the student's task, we provided a pattern program in which the students could simply add code to a set of null functions to create their own algorithms.

We also provided utility functions which they could call and a clock register which is automatically incremented to give time in milliseconds.

## SEASONGS

Dexter Morrill wrote a set of three songs which were performed by Maureen Chowing who sang and played the Radio-Baton in a new way. The baton was used not as a synthesizer controller but rather to control an effects processor which processed Maureen's voice. Typical effects were reverberation, pitch shifting, and harmonization. Program changes to change the type of effect and control changes to change a parameter in an effect (for example, reverberation time) were mapped onto movements of the batons.

### Conclusions

The pattern.c environment seems congenial for writing compositional algorithms in which the music is controlled by a combination of computer programs, random numbers generated in the computer, and motions of the batons. The C language is a very general way to write such programs. The students were able to master the parts of C that they needed much more quickly than we had expected.

The compositional algorithm environment contrasts strongly with the Conductor Program environment. Each environment has advantages and limitations. I hope the next step will be a system in which Conductor Program scores can be easily combined with compositional algorithms.

### Notes

The full text of the article is available from CCRMA.

## Optimal Signal Processing for Acoustical Systems

Bill Putnam

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

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

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

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

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

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

## 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).

## 6.5 Psychoacoustics and Cognitive Psychology

### Distance of Sound in Reverberant Fields

Jan Chomyszyn

The creation of convincing auditory perspective is an important element of computer music; it makes the sound lively and expressive. Many factors contribute to the impression of space and the location of sound sources, including appropriate reverberation, and balance of loudness and timbres of the sounds used in the composition. Some of the parameters which provide cues to distance of the sound sources are correlated in a natural reverberant environment. A typical example is direct-to-reverberant sound energy ratio and intensity, which change reciprocally along the physical distance between the sound source and the listener.

However, the percepts arising from the physical cues do not always follow the same relationship. This is easy to show in the visual world, in the case of size constancy. In visual perspective, to preserve the impression of size constancy of an object, the physical size of the object has in fact to be diminished in proportion to the provided perspective. Is this also the case in auditory perspective? Existing evidence seems to confirm this thesis.

Since the beginning of this century researchers are aware that changes in loudness and changes in distance may sometimes form equivalent concepts for the listeners (Gamble 1909). As a part of his "physical correlate theory," Warren noticed that loudness judgements of his stimuli (speech) depended on the degree of reverberation (Warren 1973). Recently Chowning (Chowning 1990) observed that loudness constancy takes place in room environment in a way analogous to the size constancy in vision. An experiment being carried out investigates this postulate with regard to computer music.

Dry percussive sounds have been produced by using the physical model of a hammer (Van Duyne 1994), simulating varying effort of the player. Next the sounds have been next reverberated at a level corresponding to changing distance in a room. In the test, subjects match dry prototypes to each of the reverberated sounds. They are also being given an auditory perspective of the room before each trial. Care has been taken to eliminate possible influence of the spectral bandwidth on the loudness match. The test will reveal if distant sounds played with greater effort are perceived as louder, as if the loudness were estimated at the sound source.

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# Embedded Pitch Spaces an The Question of Chroma: An Experimental Approach

Enrique Moreno

Since 1960, computer control of fundamental pitch has opened the door to musical experiments with arbitrary intonation systems. Researchers like M. Mathews and J. R. Pierce have investigated the psychoacoustical and musical implications of some possible synthetic intonation systems. In my book, *Expanded Tunings In Contemporary Music*, the class of all equally-tempered intonation systems is examined from several points of view (mathematical, psychoacoustical, compositional, etc.).

It was found that among the infinitely many possible intonation systems, there is one especially interesting subclass case designated as "expanded tunings." Expanded tunings are equal systems where the prime interval of the tuning, corresponding to its exponential base, is an interval of the harmonic series other than the unison or the octave and its multiples (intervals termed by J.R. Pierce as "Superconsonant Ratios").

It was also found that the perceptual and cognitive coherence of a tuning system depends on the perception of octave similarity, or pitch class abstraction, and that expanded tunings, being based not on octaves but on other superconsonant ratios, could not be perceptually coherent. These would be purely theoretical entities unless the human perceptual/cognitive apparatus could perceive or be trained to perceive prime interval similarity, or pitch class abstraction, in a way similar to that which is obtained by octaves in octave-based tunings.

It was finally found that current explanations for octave similarity support the theoretical possibility that other superconsonant ratios could elicit similar responses of similarity perception under certain special structural and musical circumstances, and provided with a meaningful musical context.

This conclusion pointed to the next step of this research by establishing the need for carefully acquired experimental evidence that will help musicians and music researchers to understand the problem of whether the human mind is capable of expanding its perceptual and cognitive abilities in finding pitch-class similarity beyond the octave.

This research examines both theoretical and empirical backgrounds to the problem, and provides an original theoretical model (based on R. Shepard's cognitive models) in order to establish an appropriate framework for conclusions. A series of cognitive experiments, their methodology, assumptions and expected interpretations as well as the techniques used to obtain results are being presently researched.

This research will contribute towards a renewed awareness of the relation between tuning schemata and music cognition, and in particular to the cognitive coherence of theoretical tuning systems in a real musical context. The proposed new model will hopefully provide an improved way to understand these relationships. Finally, it is expected that the proposed set of experiments will provide answers that contribute to our understanding of the main issue of this research, i.e., whether human cognitive mechanisms can be expected to perceive or trained to perceive, under restricted musical circumstances, similarities other than the octave with the same cognitive effects as those elicited by the octave.

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## Pitch Perception

John Pierce

The writer was invited to give a talk for "Fletcher Day" (2 June 1995) at the Washington D.C. meeting of the Acoustical Society of America, 30 May - 3 June, 1995. After examining Fletcher's publications, he chose to talk on Fletcher's discoveries concerning pitch. This topic finds no place in Fletcher's book *Speech and Hearing in Communication* (1953), recently republished by the Acoustical Society of America, and seems little known. The talk will be published in the *Journal of the Acoustical Society of America.*

Summarizing briefly, in papers in the *Physical Review* in 1924, "The Physical Criterion for Determining the Pitch of a Tone," *Phys. Rev.* 23(3), 427-437, and in "Some Further Experiments on the Pitch of Musical Tones," *Phys. Rev.* 23, 117-118, Fletcher showed that actual musical tones and synthesized tones can have the pitch of the fundamental in absence of any component of the fundamental frequency. Further, Fletcher found that pitch in the absence of the fundamental occurs only when three successive harmonics are present.

The discovery of pitch in the absence of the fundamental is sometimes attributed to J.F. Schouten, "The Residue, a New Component of Subjective Sound Analysis," *K. Ned. Akad. Wet, Proc.* 43, 356-465, 1940. Schouten's work and those following him treats pitch in the absence of the fundamental as a separate phenomenon, rather than a common characteristic of pitchiness. Fletcher's work on pitch was completed by the time the Schouten published.

Fletcher's papers on pitch have led to changes in the parts of the treatment of pitch in a book being prepared for publication by Perry Cook, based on lectures given as a part of Music 151, Psychophysics and Cognitive Psychology for Musicians.

Further consideration of Fletcher and pitch continues.

## New and Revised Psychoacoustics Textbook

John Pierce

The lectures of CCRMA's Music 151 course, "Psychophysics and Cognitive Psychology for Musicians" have supplied text and musical examples to Perry Cook, who will incorporate them into a book to be published with sound examples on compact disc. The lecturers include John Chowning, Max Mathews, Roger Shepard, Perry Cook.

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

# Psychological Representation of English Vowel Sounds

Roger Shepard, Perry Cook, and Daniel Levitin

Human speech is composed of elementary perceptual units called phonemes. These are the smallest units that we recognize as separately producible speech sounds, the vowels and consonants of our alphabet. Great effort has been focussed over the last three decades on designing sophisticated devices for the recognition and production of speech. A better understanding of how humans perceive speech might lead to advances in computer speech systems, in addition to expanding our knowledge of the human mind.

This research focuses on how humans perceive English vowel sounds, and we hope to "map" the psychological structure of vowel sounds in a way that reveals this underlying structure. The "map" will be a multi-dimensional space in which vowels that are perceived as similar will appear close together, and those that are perceived as dissimilar will appear far apart. Previous solutions have been proposed based on confusion data (Peterson and Barney; Shepard), articulatory models of speech perception (Liberman) and acoustic or spectral analysis of the speech signal.

In a new approach, we are employing an apparent motion paradigm to order the vowel sounds in similarity space.

Apparent motion is a perceptual illusion that occurs when stationary objects appear to move, due to limitations in the processing speed of the brain. For example, suppose we have two lights that are a particular distance from one another, and that flicker on and off at a particular rate. An illusion of motion is created, wherein one light appears to be moving to the position of the other. This phenomenon is the basis of motion pictures (which are comprised of a sequence of still frames) and of movie theater and Las Vegas nightclub marquis (wherein lightbulbs seem to form a moving pattern).

Apparent motion unfolds according to a predictable schedule of stages:

1. At slow alternation rates between the light sources, no motion is perceived, and the lights appear simply to alternate.

2. At more rapid alternation rates, apparent motion is perceived.

3. At still more rapid alternation rates, apparent motion breaks down, and a different perceptual event occurs, known as visual stream segregation. During visual stream segregation, people perceive the two lights as flashing on an off rapidly in two separate "streams"; no motion is perceived between them, and they do not seem to alternate.

It has been known for nearly a century that the alternation rates required for apparent motion to occur—and the alternation rates at which it breaks down into stream segregation—are linearly related to the distance between the objects. It is this relation between time and distance (Korte's Third Law of Apparent Motion) that provides the rationale for the present study.

By presenting pairs of vowel sounds in rapid alternation, we will discover the time at which steam segregation occurs. This can be used as a measure of distance between the vowel sounds.

Preliminary results suggest that the technique is valid, and that a representation of the psychological structure of speech sounds—at least according to this method—is attainable. What remains to be seen is whether the structure obtained by this method corresponds to those obtained by other methods.

# Applying Pychoacoustic Phenomena to the Coordination of Large Speaker Arrays

Steven Trautmann

Since the earliest loudspeakers, there has been a trend to create systems that "sound better" than their predecessors. While this has included all aspects of recording and reproduction of sound, a significant aspect has been using speakers in a coordinated fashion to give the illusion of localized sound sources. The work of Chowning, Bosi, and Lopez-Lezcano at CCRMA has resulted in an effective means of creating illusory localized sources on four or more channel systems. However, further improvements can be made by controlling the exact timing and content of the signals sent to speakers at known locations in a known environment since the pressure fluctuations in space are then a determined quantity.

By using many speakers, increasingly better simulations are possible. In general, N speakers can exactly reproduce a signal generated by a localized sound source at N selected points in space if the speakers are coordinated properly. This coordination is done by solving a system of linear equations to design the proper filters. By using a least squares or some other appropriate method, the pressure fluctuations at a large number of points can be approximated with a smaller number of speakers. Choosing these points in a spatial sampling pattern allows calculation of how many speakers are needed for a given a situation and error limit over the entire space. Room acoustics can cause problems, so the system would work best in a non-reverberant environment.

Trade-offs between magnitude error and phase error are also possible, allowing further refinements utilizing psychoacoustics. It is hoped that, by taking advantage of masking phenomenon, the perception of phase and magnitude, as well as head and ear acoustics, good results will be possible requiring fewer speakers and therefore potentially less overall processing.

## 6.6 Computer Music and Humanities

### The Center for Computer Assisted Research in the Humanities

CCARH

The Center for Computer Assisted Research in the Humanities (CCARH)[16] is concerned with the development of data resources for applications in music research and allied areas of humanities study.

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).

MuseData(TM): The CCARH Databases[17]

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 databases 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.

An important corollary to the development of the databases is the assembly of documentation for using the various application-specific codes. The Handbook of Musical Codes , a project initiated under the auspices of the International Musicological Society, is expected to be completed in 1995.

Contact CCARH to obtain the latest Catalogue of Works Available[18].

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[19]. In addition to editing CM, Walter B. Hewlett, the director of CCARH, and Eleanor Selfridge-Field, the resident musicologist, interact with various professional groups to further the aims of the Center in facilitating the development of a sophisticated user community. Projects in progress include the completion of the Handbook of Musical Codes[20] (with the IMS Study Group on Musical Data and Computer Applications) and the co-publication on CD-ROM of the handbook of Musical Information in Desktop Publishing[21] with the IEEE Computer Society Task Force on Computer Generated Music.

CCARH also now offers David Huron's manual UNIX Tools for Music Research: The Humdrum Toolkit[22] Reference Manual. This 551-page softbound manual describes the 60 software tools available for music

[16] http://www-ccrma.stanford.edu/CCARH/Welcome.html
[17] http://www-ccrma.stanford.edu/CCARH/musedata.html
[18] http://www-ccrma.stanford.edu/CCARH/catalogue.html
[19] http://www-ccrma.stanford.edu/CCARH/CM.html
[20] http://www-ccrma.stanford.edu/CCARH/handbook.html
[21] http://www-ccrma.stanford.edu/CCARH/desktop_pub.html
[22] http://www-ccrma.stanford.edu/CCARH/humdrum.html

research applications. These applications work with data in the Kern code. The ISBN is 0- 936943-10-6. The Toolkit itself is available by license from CCARH. Enquiries concerning UNIX Tools should be addressed to Nancy Solomon (ccarh@netcom.com), in both cases at Center for Computer Assisted Research in the Humanities 525 Middlefield Road, Ste. 120 Menlo Park, CA 94025-3443, USA (415) 322-7050; (415) 322-3307

## The International Digital Electroacoustic Music Archive

Prof. Max V. Mathews and Marcia L. Bauman

The Center for Arts and Media Technology, Karlsruhe (ZKM) and Stanford University's Center for Computer Research in Music and Acoustics (CCRMA) are pleased to announce the completion of the International Digital Electroacoustic Music Archive (IDEAMA) target collection. Co-founded by CCRMA and ZKM in December, 1990, the IDEAMA was created to collect, preserve and disseminate historically significant electroacoustic music.

An international advisory board of renowned composers was formed to help establish the international scope and reputation of the archive. To identify, locate and choose materials for the target collection, CCRMA and ZKM each formed a selection committee comprised of eminent composers, musicologists and other individuals who are well-versed and active in the field.

CCRMA and ZKM have been jointly responsible for collecting archive materials on a regional basis: ZKM has focused on European electroacoustic music, while CCRMA is responsible for music from the Americas, Asia and Australia. The original analog tapes for targeted works, composed between 1929 and 1970, have existed in a number of archives, radio stations, studios and private collections. Over 700 works have been collected and processed to form the IDEAMA target collection.

Sources for the European works include numerous major centers such as INA/GRM Paris; WDR Kln; EMS Stockholm, Experimental Studio Warszaw and the former Studio di Fonologia , Milano. In addition, works from smaller studios and private collections, and from the estate of Hermann Heiss have been included.

Sources for works in the USA include the Columbia-Princeton Electronic Music Center, the defunct San Francisco Tape Music Center (works now housed at Mills College, the University of California at San Diego, and by individual composers), the University of Illinois Experimental Music Center, Bell Telephone Laboratories (personal collection of Max Mathews), various individual composers, and commercial CDs. The Laboratoris de Investigacion y Produccion Musical (LIPM), the first major center for Latin American electroacoustic music, digitized approximately 30 works for the target collection. Japanese works have been provided by the National Center for Science Information Systems and by Dr. Emmanuelle Loubet. Most of the Japanese works were originally produced at the Tokyo studio of NHK radio.

This historic collection will be distributed to IDEAMA institutions on approximately 100 stereo audio CDs, with information about the music on the commercial FilemakerPro database. There are three types of IDEAMA institutions: founding institutions (ZKM and CCRMA), partner and affiliate institutions. The founding institutions have collaborated to establish policies and procedures for creating the archive and its ongoing function. The partner institutions have participated in the formation of the archive, in most cases by contributing materials. They will house the archive, as will new affiliate institutions.

The IDEAMA will be distributed to affiliate institutions after April, 1996. In order to become an IDEAMA affiliate institution, and for all other information, please contact Thomas Gerwin at:

ZKM/Zentrum fuer Kunst und Medientechnologie Mediathek, Kaiserstrasse 127 D-76133 Karlsruhe, Germany Phone: +49-721/9340-221 Fax: +49-721/9340-29 e-mail: tg@zkm.de

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# The Catgut Musical Acoustics Research Library

Max Mathews and Gary Scavone

## Background and History

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

The Catgut Acoustical Society (CAS) was established in 1963 to assemble available knowledge and bring together researchers in string instrument acoustics. It is best known for its pioneering work in the research and application of scientific principles to the making of conventional and new instruments of the violin family. The organization's founder, Carleen Hutchins, is regarded as the world's foremost violin maker and as an expert on the acoustics of the instrument. She has personally made more than 400 instruments of the violin family.

The Catgut Musical Acoustics Research Library makes up nearly 50 file drawers of information, including published papers, correspondences, meeting reports, extensive files on such people as Louis Condax, John Backus, Robert Fryxell, and Felix Savart, a complete set of the CAS JOURNAL publications, and two Benchmark volumes of definitive papers in violin acoustics.

CCRMA has been working to supplement the CAS files with the personal archives of Arthur Benade and John Backus. Benade was a physicist working at Case Institute of Technology and Backus was a physicist working at the University of Southern California. Both were world leaders in the acoustics of wind instruments. The personal files of John Backus were recently acquired and are now available at CCRMA. A verbal agreement for the transfer of Benade's files has been received, though a definite time frame has yet to be established.

## Purpose and Goals

The Catgut Musical Acoustics Research Library at Stanford University has been established for the purpose of preserving and maintaining a complete and up-to-date repository of knowledge on musical acoustics. CCRMA is committed to honoring requests for information from interested parties all over the world, and further to incorporate new knowledge as it becomes available. It is intended that the archive be made as accessible as possible and its existence be made known around the world.

## Activities

The process of transferring the CAS library to CCRMA and cataloging its contents is currently underway. A World Wide Web site[23] has been established to provide global access to the library (see the CCRMA home page, http://www-ccrma.stanford.edu/Welcome.html). A detailed listing of the library contents is available in this way, while requests for further information can be made via email.

[23] http://www-ccrma.stanford.edu/CCRMA/Collections/Catgut

## Impact of MIDI on Electroacoustic Art Music in the mid-1980s

Alex Igoudin

The development of electroacoustic music over the last 50 years went hand in hand with the development of technology. However, no matter how interesting research and composition efforts had been, electroacoustic composition remained on the margins of modern art music until 1980s.

The introduction of MIDI in 1984 started the technological revolution in electroacoustic instruments that brought about a fast and profound change in the tools employed by the composers for the creation of music. Compared to similar events in the history of art, the case of the standardization of digital musical instruments and computer in the mid-1980s stands out as an example of change of greater dimensions. The flood of MIDI-based hardware and software appearing within 2 years after the introduction of the standard transformed the concepts of contemporary electronic music studio, the digital instrument, and the role the computer plays for musical composition. Simultaneously, MIDI made the tools for electroacoustic composition available to a much broader audience, easing accessibility of tools to artists and performers with non-technical backgrounds, raising visibility and influence of electroacoustic composition from the margins to the front of art music.

Although much has been written about the MIDI protocol itself, virtually no available printed sources attempted to analyze what happened to the music realized with and without MIDI-based composition tools. This project aims to analyze the musicological properties of electroacoustic music in Europe and North America as they evolve under the influence of the technological revolution in the composition tools. The constituent parts of Western musical thinking, studied in traditional Musicology, transformed but nonetheless present in electroacoustic music, are brought to the fore of the current research.

Data collection for the project is based upon interviews conducted with the composers active in the electroacoustic art music before and after the introduction of MIDI. The relative closeness to the events on the music history scale brings out the particular significance of this study: accumulation of highly valuable documentary information and the rare chance to communicate with the creators of the music and to collect the research data directly from them. On the other hand, the 13-year distance from the introduction of MIDI is a long enough period to make the evaluation of events general and objective.

The interviews have already been and will be conducted in person, and by e-mail, locally in the Bay Area, nationally, and internationally. For example, a number of subjects have been interviewed in person at California College for the Arts, San Jose State University, Stanford University, University of California at Berkeley, University of California at Santa Barbara, University of California at San Diego.

## The Chorister-Chorister Interaction: an Ethnography

Paul von Hippel

Most university and conservatories nowadays teach sight-singing according to the familiar classroom loop that ethnographers have called "initiate-respond-evaluate" (IRE): the teacher asks a student to sing a particular exercise, the student tries to, and the teacher evaluates the student's effort.

IRE is excellent at displaying how well individual students can read the exercises. Since schools are largely in the business of sorting students by ability, it is not surpising that such a method has grown pervasive. As

a way for students to learn, however, IRE is rather tedious and wasteful. Students spend most of their time watching others being evaluated.

Outside the classroom, people learn to sight-sing by participating in choruses. (Some choruses, of course, encourage the skill more than others.) In a chorus, singers learn not only from the conductor, but from each other. A chorus, in fact, might be understood as a metaphor for collaborative learning. Little work, unfortunately, has been done on how choristers pick things up from one another.

I am presently conducting an ethnography of one singer's experience in a sight-singing class that was run as a choir. I am collecting data through a video camera at the front of the room, and through a microphone clipped to the singer's lapel. In analyzing the data, I will focus on moments when the singer deviates from the singers around her. I am interested in two questions: first, under what circumstances does she interpret her deviation as an error; second, under what circumstances does she correct herself?

## 6.7 Past Research Projects

The following research summaries are contained in Wood and Van Duyne, "Overview, September 1994," CCRMA Report 1994, Stanford Department of Music Technical Report No. STAN-M-97.

COMPUTER MUSIC SOFTWARE DEVELOPMENT

**The CCRMA Music Kit and DSP Tools Distribution** David Jaffe, Julius Smith

**Rapid Prototyping of Digital Signal Processing and Synthesis Systems** Julius Smith

**The Design of SynthBuilder — A Graphical SynthPatch Development Environment for the NeXTSTEP** Nick Porcaro

**Common LISP Music and Common Music Notation** William Schoettstadt

**A Dynamic Spatial Sound Movement Toolkit** Fernando Lopez-Lezcano

**Common Music / Stella: A Music Composition Language in Common LISP and CLOS** Heinrich Taube

**DMIX** Daniel V. Oppenheim

PHYSICAL MODELING AND DIGITAL SIGNAL PROCESSING

**Digital Waveguide Modeling of Acoustic Systems** Julius O. Smith

**Waveguide Reverberators and Real-Time Implementations** R.J.Fleck

**Feedback Delay Networks** Davide Rochesso

**The 2-D Digital Waveguide Mesh** Scott Van Duyne, Julius O.Smith

**The Wave Digital Hammer** Scott Van Duyne, Julius O. Smith

**The Haptically Controlled Waveguide Piano** Julius O. Smith, Scott Van Duyne, Brent Gillespie

**Synthesis of the Singing Voice Using a Physically Parameterized Model of the Human Vocal Tract** Perry R. Cook

**Adding Pulsed Noise to Wind Instrument Physical Models** Chris Chafe

**Physical Modeling of Brasses** David Berners

**Synthesis and Research on Reed Driven Woodwind Instruments with Particular Emphasis on the Saxophone** Gary P. Scavone

**A Passive Nonlinear Filter for Physical Models** John R.Pierce, Scott Van Duyne

**Physical Modeling of Music Instruments Using Neural Networks** Wen-Yu Su

**FFT-Based Signal Processing and Spectral Modeling Synthesis** Julius O. Smith

**Instantaneous and Frequency-Warped Signal Processing Techniques for Pitch Tracking and Source Separation and Fast Linear-Phase FIR Filter Theory and Design** Avery Wang

**Time-Frequency Analysis of Audio** Scott Levine

CONTROLLERS FOR COMPUTER AND MUSICAL INSTRUMENTS

**Real-Time Controllers for Physical Models** Chris Chafe

**Ongoing Work in Brass Instrument Synthesizer Controllers** Perry R. Cook, Dexter Morrill

**Performer-Oriented Brass Instrument Synthesis and Control** Timothy Stilson

**Haptic User Interfaces for the Blind** Sile O'Modhrain, Brent Gillespie

**The Touchback Keyboard** Brent Gillespie

**Progress Report on the Radio Baton and Conductor Program** Max V. Mathews

**PadMaster: An Improvisation Environment for the Radio Drum** Fernando Lopez Lezcano

**The Computer-Extended Ensemble** David A. Jaffe

**Real-Time Control Using Biological Signals** Bill Putnam

**Biocontrol Interfaces as Musical Instruments** Atau Tanaka

## PSYCHOACOUSTICS AND COGNITIVE PSYCHOLOGY

**Applying Psychoacoustic Phenomenon to the Coordination of Large Speaker Arrays** Steven Trautmann

**Distance of Sound in Reverberant Fields** Jan Chomyszyn

**Pitch and Repetition Rate Perception** John R. Pierce

**Embedded Pitch Spaces and the Question of Chroma: An Experimental Approach** Enrique I. Moreno

**Musical Perception and Memory** Daniel J. Levitin

**Statistical Models for Psychoacoustic Research** Daniel J. Levitin

**Psychological Representation of English Vowel Sounds** Roger Shepard, Perry R. Cook, Daniel J. Levitin

**The Perceptual Organization of Sound** Albert S. Bregman

**New and Revised Psychoacoustic Textbooks** John R. Pierce, et al

**Psychoacoustic Textbooks** John R. Pierce, et al

# Chapter 7

# Publications

For publication ordering please contact CCRMA (see p.4).

## 7.1 Publications


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