Digital Rock and Roll: Implementation of the SoundDroid Fine Editor

John Strawn
The Droid Works
P. O. Box CS-8180
San Rafael CA 94912, USA
Tel. (415) 485 5000

Several signal processing modules inspired in part by traditional analog techniques have been developed for the SoundDroid. One of these is the Fine Editor, abbreviated FinEd (pronounced "Fine, Ed"), whose specifications were developed by Peter Nye of The Droid Works, the staff of Sprocket Systems at Lucasfilm, and the author. This implementation borrowed heavily from a module implemented by James A. Moorer of The Droid Works and used in creating arrow sounds for *Indiana Jones and the Temple of Doom.* At 166 microinstructions, FinEd is the largest and most complicated module yet written for the SoundDroid.

Part of the inspiration for FinEd was the desire to be able to digitally emulate "tape rock and roll" (hence the title of this paper). That is, the engineer rocks the tape reels of an analog tape deck back and forth while maintaining the tension of the tape on the playback head. This is very useful for isolating, say, the beginning of a recording. The SoundDroid's video monitor, which shows a video clip of the tape for which sound effects are being edited, can step forwards or backwards through the video clip under control of a "shuttle knob." The FinEd module makes the sound follow what is displayed on the video monitor.

In editing cues and sound effects, it is also necessary to be able to isolate part of a recording, fade into and out from the selected points, and possibly loop through the selected passage. In looping, a crossfade may be needed as playback wraps around from the end of the recording to the beginning; or a user-specified pause may intervene between the end and the beginning. It may be necessary to speed up or slow down the looped recording. FinEd contains these capabilities as well.

The theme of this paper is not to discuss why such features are needed or how such features are used in a commercial production setting, but rather to cover some details of the implementation that may be of interest to others.

The Audio Signal Processor (ASP) is the numerical heart of the SoundDroid. Each ASP contains two banks of "bulk memory," with each bank holding 16 M words of 24-bit memory. Each bank can be arbitrarily partitioned into buffers ranging in size from 512 words to 16 M words. In the recording part of FinEd, sound samples are recorded into sequential locations in a buffer in this memory. Playing back the signal requires that FinEd step through the buffer in memory. For playback at normal speed, an increment of 1 is used. Setting the increment to 2 plays the sound back twice as fast. Typically, a buffer of length 256 K words is used. Since the ASP has 24-bit internal busses, and since it takes only 18 bits to represent the addresses for the 256 K word buffer, the 6 low-order bits of the incremented address can be used for fractional increments. Thus, setting the increment to 0.5 plays the sound back twice as slow, and so on, with the slowest speed being $1/2^6 = 1/64$th as slow as the original. (This changes, of course, if other buffer lengths or double-precision addressing is used). For "rock and roll," the processor controlling the ASP simply changes the increment on the fly.

As we all know, various forms of aliasing and/or distortion can occur if the sample rate is arbitrarily modified in this fashion. Digital implementations of the speedup/slowdown playback on other systems have solved these problems with interpolating low-pass filters, which is computationally expensive. I was surprised to find that it was adequate to linearly interpolate, using the 6 bits mentioned above to determine the fractional part for interpolation. Aliasing and other distortion are still audible, but the effects are negligible.

During crossfade, there are not one but two pointers into the recorded signal. The first marches through the recording being modified by the fadeout, and the other follows the part being faded in. Obviously, the fadeout and fadein times must be identical during crossfade (they are independent when no crossfade occurs). Crossfading doubles the number of accesses to bulk memory; this is unfortunately expensive, as one memory access requires several microinstruction cycles (of course those cycles can be used for other operations). Also, note that when the crossfade is finished, the "main" pointer that has just finished the fadeout must be carefully reset to include any fractional part that may have been accumulated by the "secondary" pointer that was active during the fadein.

Fadein and fadeout are implemented using a single-pole filter. For the fadein we have the general form

$$y[n] = y[n-1] + G_a * (x[n] - y[n-1])$$

where G_a is a gain term. To create the desired exponential fadein, $x[n]$ is always 1.0, which can be approximated in the 24-bit number system of the ASP as 77777777_8. But $77777777_8 - y[n-1]$ is the same as the one's complement of $y[n-1]$, so the filter simplifies to

$$y[n] \approx y[n-1] + G_a * (\neg y[n-1]),$$

where "$\neg$" indicates the one's complement. This form is useful because it is more efficient to calculate $\neg y[n-1]$ in the ASP than to calculate $(1.0 - y[n-1])$. As the ASP is a horizontally microcoded machine, it is possible to be clever about coding and to do one memory fetch of $y[n-1]$ where it is needed twice. For the fadeout, we have

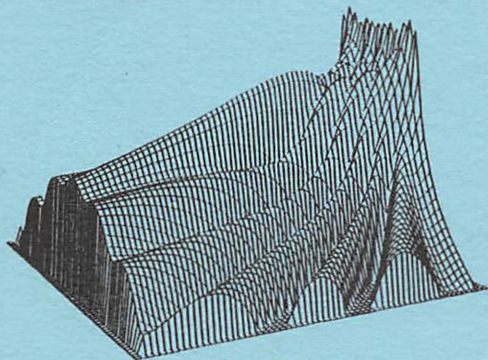
$$y[n] = y[n-1] - G_d * (y[n-1])$$

and the horizontally microcoded architecture can again be exploited. Calculating G_a and G_d is not as straightforward as it might seem; for example, these values must be changed as the effective sample rate changes.

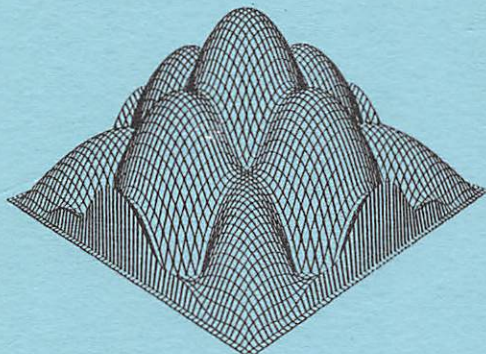
FINAL PROGRAM
AND
PAPER SUMMARIES FOR THE 1986 IEEE ASSP WORKSHOP ON APPLICATIONS OF SIGNAL PROCESSING TO AUDIO AND ACOUSTICS

September 15-17, Mohonk Mountain House
New Paltz, New York

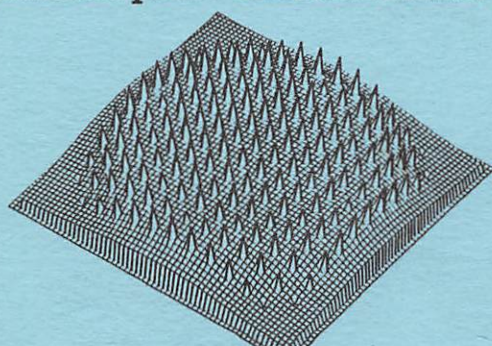
Sponsored by the
IEEE Acoustics, Speech, and Signal Processing Society



Focussed Line Array



Nearfield Response Uniform Planar Array



Nearfield Response Shaded Planar Array

WORKSHOP TECHNICAL PROGRAM

MONDAY

Plenary Talk:

8:00 am, Monday, September 15, 1986

"Sound Processing for Better Acoustics and Improved Hearing," *M. R. Schroeder, AT&T Bell Laboratories, 600 Mountain Ave., Murray Hill, NJ 07974*; and *the University of Gottingen, Gottingen, WEST GERMANY*

Session 1. Wideband Audio and Teleconferencing

8:45 - 10:30 am, Monday, September 15, 1986
Chairperson: *N. S. Jayant, AT&T Bell Laboratories, 600 Mountain Ave., Murray Hill, NJ 07974*

1.1 **"High Quality Sound Decoder Using Noise Shaping Techniques and Digital Filters,".** *G. Bars* and *Jean-Pierre Petit, Centre National d'Etudes des Telecommunications - 22301 Lannion, FRANCE*

1.2 **"A Digital Coder for Wideband Audio Applications,"** *William R. Belfield, AT&T Bell Laboratories, Holmdel, NJ 07733*

1.3 **"Digital Signal Processing in Audio Bridging Systems,"** *David R. Fischell, AT&T Bell Laboratories, Holmdel, NJ 07733*

1.4 **"A Pole-Zero Adaptive Acoustic Echo Canceller in Teleconferencing,"** *Guozhu Long, Dennis Shwed* and *David D. Falconer, Department of Systems and Computer Engineering, Carleton University, Ottawa, Ont. CANADA K1S 5B6*

1.5 **"Second-Order Toroid Microphone for Teleconferencing,"** *J. E. West* and *G.M. Sessler, Acoustics Research Department, AT&T Bell Laboratories, 600 Mountain Ave., Murray Hill, NJ 07974*

1.6 **"Computer-Steered Microphone Arrays for Large Room Teleconferencing,"** *G. W. Elko, J. L. Flanagan*, and *J. D. Johnston, Acoustics Research Department, AT&T Bell Laboratories, Murray Hill, NJ 07974*

1.7 **"The Use of Signal Processing Techniques for an FM Demultiplexer Receiver,"** *Stephanie Ronald, Tony May*, and *Grant Griffin, Rockwell - Collins, M/S 137-154, 855 35th Street N.E., Cedar Rapids, IA 52498*

Break

Session 2. Perceptual Imaging, Reconstruction and Audio Measurements

10:45 am - 12:30 pm, Monday, September 15, 1986
Chairperson: *Stanley P. Lipshitz, University of Waterloo, Waterloo, Ontario N2L 3G1, CANADA*

2.1 **"Computer Analysis of Sound Imaging in Stereo Systems,"** *Duane H. Cooper* and *Jerald L. Bauck, Dept. of Electrical and Computer Engineering, University of Illinois, 1406 W. Green St., Urbana, IL 61801*

2.2 **"The Vector-Valued Impulse Response and Its Application to Three-Dimensional Sound Field Reconstruction,"** *Anthony J. Romano, The Pennsylvania State University, Graduate Program in Acoustics, Applied Research Laboratory, P. O. Box 30, State College, PA 16804*

2.3 **"Experiments on Active Reverberation and Diffusion Control,"** *D. Guicking, M. Wenzel* and *J. Schloffel, Drittes Physikalisches Institut der Universitat Gottingen, Burgerstr. 42-44, D-3400 Gottingen, West Germany*

2.4 **"Parameter Estimation of Acoustic Models: Audio Signal Separation,"** *Erling Wold* and *Alvin M. Despain, Computer Science Division, Department of EECS, University of California, Berkeley, CA 94720*

2.5 **"A Crossover Network for a Hi-Fi Loudspeaker,"** *Svein Sorsdal* and *Sverre Stensby, The Norwegian Institute of Technology, Division of Telecommunications, N-7034 Trondheim - NTH, NORWAY*

2.6 **"The Locatability of Telephone Tone-Caller Sounds,"** *Malcolm King, Standard Telecommunication Laboratories Limited, London Road, Harlow, Essex, CM17 9NA, ENGLAND*

2.7 **"Sound Fields in Rooms as Holographic Structures of Information,"** *James B. Lee, Concert Acoustics, P.O. Box 18017, Portland, Oregon 97218*

Lunch

Session 3. Sound Editing, Enhancement and Special Effects

2:00 - 4:30 pm, Monday, September 15, 1986
Chairperson: *James A. Moorer, The Droid Works, P. O. Box CS 8180, San Rafael, CA 94912*

3.1 **"Digital Rock and Roll: Implementation of the SoundDroid Fine Editor,"** *John Strawn, The Droid Works, P.O. Box CS-8180, San Rafael, CA 94912*

3.2 **"Audio Pre-processing by Sinusoidal Analysis/Synthesis,"** *Thomas F. Quatieri, R.J. McAulay, and J.T. Lynch, Lincoln Laboratory, Massachusetts Institute of Technology, Lexington, Massachusetts 02173-0073*

3.3 **"Time Scale Modification of Signals Using A Synchronous Gabor Technique,"** *Douglas L. Jones* and *Thomas W. Parks, Dept. of Electrical and Computer Engineering, Rice University, P.O. Box 1892, Houston, TX 77251*

3.4 **"How to Find Edit Points on Digital Audio Tapes Using Multirate FIR Digital Filter Realized with the TMS320,"** *Daniele Pelloni, Willi Studer AG, Althardstrasse 30, 8105 - Regensdorf, SWITZERLAND*

3.5 **"An Optical-Disk Sound Effects Retrieval Device,"** *James A. Moorer, The Droid Works, P.O. Box CS 8180, San Rafael, CA 94912*

3.6 **"Phase Dispersion and Peak-Limiting for Dynamic Range Compression,"** *John T. Lynch, Thomas F. Quatieri*, and *R.J. McAulay, Lincoln Laboratory, Massachusetts Institute of Technology, Lexington, Massachusetts 02173-0073*

3.7 **"Noise Reduction in Speech by Linear Prediction,"** *F. Yarman-Vural* and *L. Paarmann, Department of Electrical and Computer Engineering, Drexel University, Philadelphia, PA 19104*

3.8 **"A Survey of Looping Algorithms for Sampled Data Musical Instruments,"** *Dana C. Massie, E-mu Systems, Inc., 1600 Green Hills Rd., Scotts Valley, CA 95066-4542*

3.9 **"The application of digital analysis-synthesis techniques to acoustic trainers,"** *M. A. Deaett, M/S 186, Raytheon Company, Submarine Signal Division, West Main Road, Portsmouth, R.I. 02872*

Panel Discussion

7:30 - 9:30 pm, Monday, September 15, 1986

"Control of Echo and Reverberation for Teleconferencing"
Moderator: M. M. Sondhi, *AT&T Bell Laboratories, 600 Mountain Ave., Murray Hill, NJ 07974*
Panelists:
Radamis Botros, *BOTA Consulting Ltd., Ottawa, Ontario. CANADA K1Y 3E6*
S. J. Campanella, *Comsat Laboratories, Clarkesburg, MD 20734*
D. L. Duttweiler, *AT&T Bell Laboratories, Holmdel, NJ 07733-1988*
D. Guicking, *University of Gottingen, D-3400 Gottingen, WEST GERMANY*
Kazunori Ozawa, *NEC, Kawasaki 213, JAPAN*

TUESDAY

Plenary Talk:

8:00 am, Tuesday, September 16, 1986

"Signal Processing to Aid the Hearing-Impaired"
Dr. Harry Levitt, The Graduate School and University Center of the City University of New York, 33 West 42 Street, New York, New York 10036

Session 4. Signal Processing Aids to the Handicapped

8:45 - 10:15 am, Tuesday, September 16, 1986
Chairperson: *D. M. Chabris, Electrical Engineering Department, Brigham Young University, Provo, UT 84602*

4.1 **"Computer Simulation of Hearing Aid Performance,"** *James M. Kates, Siemens Hearing Instruments, Inc., 685 Liberty Ave., Union, NJ 07083*

4.2 **"Recognition of Useful Information in the Speech of the Nonverbal,"** *J.R. Deller, Jr.*, and *D. Hsu, Digital Signal Processing Laboratory: Speech Processing Sector, Department of Electrical and Computer Engineering - 409 DA; L.J. Ferrier*, and *L.L. Hanrahan, Dept. of Speech and Language Pathology and Audiology - 106 FR; Northeastern University, 360 Huntington Ave., Boston, MA 02115;* and *H.C. Shane, Communication Enhancement Clinic, The Childrens Hospital, 300 Longwood Ave., Boston, MA 02115*

4.3 **"The Development of Devices for the Hearing-Impaired - A Progress Report: Part 1,"** *A. Maynard Engebretson, R.E. Morley, M.P. O'Connell,* and *G.L. Engel, Center Institute for the Deaf, 818 S. Euclid Ave., St. Louis, MO 63110,* and *Washington University, Dept. of Electrical Engineering, Skinker and Forsyth, St. Louis, MO 63130*

4.4 **"The Development of Devices for the Hearing-Impaired - A Progress Report: Part 2,"** *R.E. Morley, A. Maynard Engebretson, G.L. Engel,* and *M.P. O'Connell, Washington University, Dept. of Electrical Engineering, Skinker and Forsyth, St. Louis, MO 63130,* and *Center Institute for the Deaf, 818 S. Euclid Ave., St. Louis, MO 63110*

4.5 **"A Frequency Domain Digital Hearing Aid,"** *Richard W. Christiansen, R. Brey, D.M. Chabris, and M. Robinette, Electrical Engineering Department, Brigham Young University, Provo, UT 84602*

4.6 **"The AKL representation for Tactile aids for the deaf,"** *Grayson Abbott,Philip B. Bowman, Susan A. Hutchinson, and Mathew J. Miller, Creare Inc., Etna Road, P.O. Box 71, Hanover, NH 03755*

Break

+
Session 5. Music Analysis/Synthesis and New Musical Sounds

10:30 am - 12:30 pm, Tuesday, September 16, 1986
Chairperson: *Ercolino Ferretti, Department of Computer Science, The University of Utah, 3160 Merrill Engineering Building, Salt Lake City, Utah 84112*

5.1 **"Prony's Method for the Analysis and Synthesis of Musical Notes,"** *A.S. Noetzel, Dept. of EE/CS, Polytechnic Univeristy, 333 Jay Street, Brooklyn, New York 11201*

5.2 **"Digital Coding of Musical Sound - Some Statistics of Interest,"** *J. D. Johnston, AT&T Bell Labs, Murray Hill, NJ 07974*

5.3 **"Phase modulation with interpolated time functions : a powerful algorithm for sound synthesis,"** *S. Cavaliere and Aldo Piccialli, Universita Di Napoli, Dipartimento Di Fisica, Mostra d'Oltremare, Pad. 19-20 - I 80125, Napoli, ITALY*

5.4 **"A Mathematical Basis Function for Modeling Natural Sounds,"** *Ercolino Ferretti, Department of Computer Science, The University of Utah, 3160 Merrill Engineering Building, Salt Lake City, Utah 84112*

5.5 **"Use of LPC Spectral Estimation for Music: Analysis, Processing and Synthesis,"** *Xavier Rodet* and *Philippe DePalle, Groupe CHANT/FORMES, IRCAM, 31 rue Saint Merri, 75004 Paris, FRANCE*

5.6 **"Synthesis by formants: A new approach,"** *S. Cavaliere, I. Ortosecco,* and *Aldo Piccialli, Universita Di Napoli, Dipartimento Di Fisica, Mostra d'Oltremare, Pad. 19-20 - I 80125, Napoli, ITALY*

5.7 **"Efficient Simulation of the Reed-Bore and Bow-String Mechanisms,"** *Julius O. Smith, Center for Computer Research in Music and Acoustics (CCRMA), Department of Music, Stanford University, Stanford, CA 94305*

5.8 **"Study on Pole-Zero Modeling in Analysis-Synthesis of Bamboo Tone, Shakuhachi, and Its Transient,"** *Michiko Toyama, 1-29018, Hanegi, Setagaya-ku, Tokyo 156, JAPAN*

Lunch

Session 6. Signal Processing Algorithms, Hardware, and Software

2:00 - 4:30 pm, Tuesday, September 16, 1986
Chairperson: *Jonathan Allen, Massachussette Institute of Technology, Cambridge, MA 02139*

6.1 **"Structural, Algorithmic, and Time/Space Tradeoffs for a Cost Effective TMS 32010/IBM-PC-Based Realtime Digital Speech Spectrograph,"** *L. Robert Morris, System and Computer Engineering, Carleton University, Ottawa, CANADA K2C 3J1*

6.2 **"Implementation of the Stochastic Gradient Adaptive Filtering Algorithm on the ADSP-2100 Digital Signal Processor,"** *Cole Erskine, Analog Devices Inc., P.O. Box 280, Norwood, MA 02062*

6.3 **"A Fast VLSI CORDIC Algorithm Using Redundant Binary Arithmetic,"** *J.C. Bu, H.X. Lin, Ed F.A. Deprettere*, and *P. Dewilde, Delft University of Technology, Department of Electrical Engineering, Mekelweg 4, 2628 CD Delft, The NETHERLANDS*

6.4 **"PC Based Acoustic Signal Processing Test-Bed,"** *John W. Irza, The Charles Stark Draper Laboratory, Inc., 555 Technology Square, Cambridge, Massachusetts 02139*

6.5 **"DSP.*: A DSP-32 Based Multiprocessor,"** *Mark Kahrs, AT&T Bell Laboratories, Murray Hill, NJ 07974* X

6.6 **"Time-Space Optimization of TMS320 Software for Leroux-Gueguen LPC Matrix Inversion and K-to-A Parameter Transformation,"** *L. Robert Morris, System and Computer Engineering, Carleton University, Ottawa, CANADA K2C 3J1*

6.7 **"SYSID - A PC-Based Audio Frequency System Characterization Package,"** *V. Pluvinage, K. Swaminathan, C.B. Fong*, and *J.B. Allen, Rm 3G-429, AT&T Bell Laboratories, Holmdel, NJ 07733* X

6.8 **"The TMS320C25: A 100ns CMOS VLSI Digital Signal Processor,"** *Ray Simar, Jr.* and *Jay B. Reimer, P. O. Box 1443, Texas Instrument Inc., Houston, Texas 77251* X

6.9 **"Efficient Algorithms, an architecture and custom I.C.'s for digital audio signal processing,"** *J. Van Ginderdeuren, R. Govaerts, Katholieke Universiteit Leuven, Departement Elektrotechniek - Afdeling E.S.A.T., Kardinaal Mercierlaan 94, B - 3030 Heverlee*, and *H. De Man, Interuniversity Micro-Electronics Center, Leuven, BELGIUM* X

X
Panel Discussion

7:30 - 9:30 pm, Tuesday, September 16, 1986

"Architectural Viewpoints for DSP Microchips"
Moderator: L. Robert Morris, *Systems and Computer Engineering, Carleton University*, and *DSPS, Inc. Ottawa, CANADA K2C 3J1*
Panelists:
Jack Roesgen, *Analog Devices*
Mark Davis, *NEC*
Andrew G. Deczky, *Bell Northern*
Gene Frantz, *Texas Instruments*
Juan B. Schiappacasse, *National Semiconductor*

WEDNESDAY

Plenary Talk:

8:00 am, Wednesday, September 17, 1986

"Impact on Signal Processing by New Design Techniques for Custom Silicon,"
Dr. Hal Alles, Silicon Design Labs., 645 Martinsville Road, Liberty Corner, NJ 07938

Session 7. Perceptual Criteria, Audio Measurements and Evaluation

8:45 - 10:45 am, Wednesday, September 17, 1986
Chairperson: *Richard C. Cabot, Audio Precision Incorporated, P. O. Box 2209, Beaverton, OR 97075*

7.1 **"On the Use of Computer Generated Dithered Test Signals,"** *Robert A. Finger, Acoustics and Signal Processing Audio Research Group, CBS Technology Center, CBS Inc., 227 High Ridge Road, Stamford, Connecticut 06905*

7.2 **"A New High Resolution D/A Conversion System and Its Computer Evaluation",** *Masao Kasuga, Victor Company of Japan, LTD. Audio Engineering Research Center, 1766-1 Shimotsuruma, Yamato-shi, Kanagawa-ken, 242 JAPAN*

7.3 **"Quantization Noise in an Oversampled A/D Converter,"** *L. Richard Carley, Department of Electrical and Computer Engineering, Carnegie Mellon University, Schenley Park, Pittsburgh, PA 15213*

7.4 **"A New Windowing Technique for Digital Harmonic Distortion Measurement,"** *Robert W. Adams, dbx inc., 71 Chapel St., Newton, MA 02195*

7.5 **"A Distortion Measurement System Based on Auditory Modelling,"** *Matti Karjalainen, Helsinki University of Technology, Acoustic Lab., Otakaari 5A, 02150 Espoo, FINLAND*

7.6 **"3-D Modeling Tools for Accurately Predicting Loudspeaker Array Coverage Patterns,"** *David G. Meyer, School of Electrical Engineering, Purdue University, West Lafayette, IN 47907*

7.7 **"Features of Signals with Perceptible Phase Distortion,"** *D. Preis, Dept. of Electrical Engineering, Tufts University, Medford, MA 02155* and *P.J. Bloom, Division of Engineering, Polytechnic of Central London, London, ENGLAND*

7.8 **"A Physical Understanding of Acoustic Intensity and Application to Transducer Radiation,"** *Jiri Tichy* and *J. Adin Mann, Graduate Program in Acoustics, The Pennsylvania State University, P.O. Box 30, State College, PA 16804*

7.9 **"A New Cepstral Decomposition for Electroacoustic Transducer Measurement,"** *Stanley P. Lipshitz and John Vanderkooy, Audio Research Group, University of Waterloo, Waterloo, Ontario N2L 3G1, CANADA,* and *Paul Bauman, Communications Research Laboratory, McMaster University, Hamilton, Ontario L8S 4K1, CANADA*

Break

Session 8. General Discussion

11:00 - 12:00 am, Wednesday, September 17, 1986
All participants of the workshop

Lunch

WORKSHOP SCHEDULE			
DAY	TIME	PARLOR	OTHER
Sunday	4:00pm 6:30pm		Registration: Lake Lounge
	6:30pm 8:00pm		Dinner: Main Dining Room
	8:00pm 10:00pm		Cash Bar: West Dining Room
Monday	7:00am 8:00am		Registration: Lake Lounge Breakfast: Main Dining Room
	8:00am 8:45am	Welcome Address Talk: Sound Processing	
	8:45am 10:30am	Wideband Audio and Teleconferencing	
	10:30am 10:45am		Coffee Break: West Alcove Registration: Lake Lounge
	10:45am 12:30pm	Perceptual Imaging and Reconstruction	
	12:30pm 1:30pm		Lunch: Main Dining Room
	2:00pm 4:30pm	Sound Editing and Enhancement	
	6:30pm 7:30pm		Dinner: West Dining Room
	7:30pm 9:30pm	Panel Discussion: Noise & Echo Control	
Tuesday	7:00am 8:00am		Breakfast: Main Dining Room
	8:00am 8:45am	Talk: Aids to the Hearing-Impaired	
	8:45am 10:15am	Aids to the Handicapped	
	10:15am 10:30am		Coffee Break: West Alcove
	10:30am 12:30pm	Music Analysis New Music Sounds	
	12:30pm 1:30pm		Lunch: Main Dining Room
	2:00pm 4:30pm	Signal Processing Algor, Hdwr, & Sftwr	
	6:30pm 7:30pm		Dinner: West Dining Room
	7:30pm 9:30pm	Panel Discussion DSP Architectures	
Wednesday	7:00am 8:00am		Breakfast: Main Dining Room
	8:00am 8:45am	Talk: Designing Custom Silicon	
	8:45am 10:45am	Audio Measurements	
	11:00am 12:00am	General Discussion	
	12:30pm 1:30pm		Lunch: Main Dining Room