Data Sonification and Sound Visualization

Hans G. Kaper

Mathematics and Computer Science Division, Argonne National Laboratory

Sever Tipei

School of Music, University of Illinois

Elizabeth Wiebel

Mathematics and Computer Science Division, Argonne National Laboratory

Abstract

This article describes a collaborative project between researchers in the Mathematics and Computer Science Division at Argonne National Laboratory and the Computer Music Project of the University of Illinois at Urbana-Champaign. The project focuses on the use of sound for the exploration and analysis of complex data sets in scientific computing. The article addresses digital sound synthesis in the context of DIASS, a Digital Instrument for Additive Sound Synthesis, and sound visualization in a virtual-reality environment by means of M4CAVE. It describes the procedures and preliminary results of some experiments in scientific sonification and sound visualization.

While most computational scientists routinely use visual imaging techniques to explore and analyze large data sets, they tend to be much less familiar with the use of sound. Yet, sound signals carry significant amounts of information and can be used advantageously to increase the bandwidth of the human/computer interface. The project described in this article focuses on scientific sonification—the faithful rendering of scientific data in sounds—and the visualization of sounds in a virtual-reality environment. The project, which grew out of an effort to apply the latest supercomputing technology to the process of music composition (see Box 1), is a joint collaboration between Argonne National Laboratory (ANL, Mathematics and Computer Science Division) and the University of Illinois at Urbana-Champaign (UIUC, Computer Music Project).

Digital sound synthesis is addressed in Section 1. The discussion is centered around DIASS, a Digital Instrument for Additive Sound Synthesis. Section 2 describes some experiments in scientific sonification. Sound visualization in a virtual-reality (VR) environment is discussed in Section 3. Here, the main tool is M4CAVE, a program to visualize sounds from a score file. Section 4 contains some more general observations about the project.

1 Digital Sound Synthesis

Digital sound synthesis is a way to generate a stream of numbers representing the sampled values of an audio waveform. To realize the sounds, one sends these samples through a digital-to-analog converter (DAC), which converts the numbers to a continuously varying voltage that can be amplified and sent to a loudspeaker.

One way of viewing the digital sound-synthesis process is to imagine a computer program that calculates the sample values according to a mathematical formula and sends those samples, one after the other, to the DAC. All the calculations are carried out by a program, which can be changed in arbitrary ways by the user. From this point of view, digital synthesis is the same as software synthesis. Software synthesis contrasts with hardware synthesis, where the calculations are carried out in special circuitry. Hardware synthesis has the advantage of high-speed operation, but lacks the flexibility of software synthesis. Software synthesis is the technique of choice if one wishes to develop an instrument for data sonification.

With software synthesis, one can indeed realize any imaginable sound—provided one has the time to wait for the results. With a sampling rate of 44,100 samples per second the time available per sample is only 20 microseconds, too short for real-time synthesis of reasonably complex sounds. For this reason, most of today's synthesis programs generate a sound file, which is then played through a DAC. But data sonification in real time may become feasible on tomorrow's high-performance computing architectures. Our research effort focuses on the development of a flexible and powerful digital instrument for scientific sonification and on finding optimal ways to convey information through the medium of sound.

1.1 DIASS – A Digital Instrument

Two pieces of software consitute the main tools of the project: DIASS, a Digital Instrument for Additive Sound Synthesis, and M4CAVE, a program for the visualization of sound objects in a multimedia environment. Both are part of a comprehensive *Environment for Music Composition*, which includes additional software for computer-assisted composition and automatic music notation. Figure 1 gives a schematic overview of the various elements of the *Environment*; C and S mark the data entry points for composition and sonification, respectively.

In this section we describe the workings of DIASS; we will describe M4CAVE after we have discussed our ideas on scientific sonification.

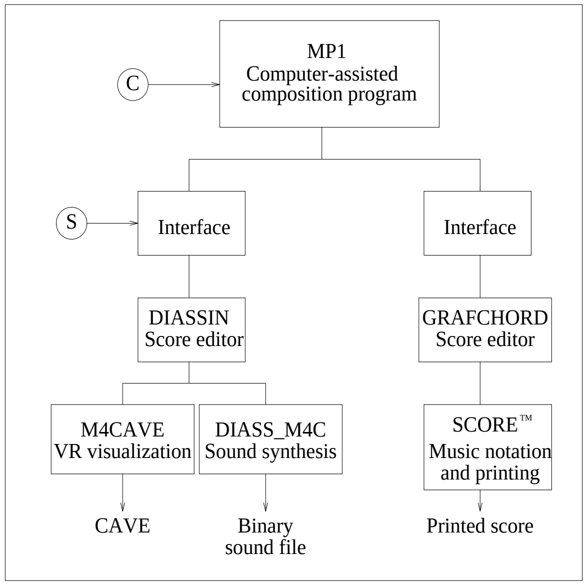


Figure 1: The *Environment for Music Composition*.

1.1.1 The Instrument

The DIASS instrument functions as part of the M4C synthesis language developed by Beauchamp and his associates at the University of Illinois [3]. Synthesis languages like M4C are designed around the notion that the user creates an instrument together with a score that references the instrument. The synthesis program reads the instrument, feeds it the data from the score file, and computes the final audio signal, which is then written to a sound file for later playback [16].

The M4C synthesis language is imbedded in the C language. As part of the current project, the instrument and relevant parts of M4C were redesigned for a distributed-memory environment. The parallel implementation uses the standard MPI message-passing library [6].

Like all additive-synthesis instruments, DIASS creates sounds through a summation of simple sine waves, so the basic formula is

$$S(t) = \sum_i P_i(t) = \sum_i a_i(t) \sin(2\pi f_i(t)t + \phi_i(t)).$$

The individual sine waves which make up a sound are commonly designated as the "partials" of the sound; hence, the symbol P. The sum extends over all partials that are active at the time t; a_i is the amplitude, f_i the frequency, and ϕ_i the phase of the ith partial. These variables can be modulated periodically or otherwise; the modulations evolve on a slow time scale, typically on the order of the duration of a sound. Phase modulation is barely distinguishable from frequency modulation, particularly in the case of time-varying frequency spectra, and is not implemented in DIASS.

The range of audible frequencies ranges roughly from 20 to 20,000 Hz, although in practice the upper limit is one-half the sampling frequency (Nyquist criterion).

The partials in a sound need not be in any harmonic relationship (that is, f_i need not be a multiple of some fundamental frequency f_0), nor do they need to share any other property. The definition of a sound is purely operational. What distinguishes one "sound" from another is that certain operations are defined at the level of a sound and affect all the partials that make up the sound.

The evolution of a partial can be subject to many other controls, besides amplitude and frequency modulation. Moreover, these controls can affect a single partial or all the partials in a sound. For example, reverberation, which represents the combined effects of the size and acoustic characteristics of the hall, affects all the partials in a sound simultaneously, although not necessarily in the same way. Furthermore, if a random element is present, it must be applied at the level of a sound; otherwise, a complex wave is perceived as a collection of independent sine waves, instead of a single sound. Hence, it is important that all partials in a sound access the same random number sequence and that the controls of any partial that changes its allegiance and moves from one sound to another be adjusted accordingly.

Table 1 lists the control parameters that can be applied in DIASS. Some, like starting time and duration, do not change for the duration of a sound; they are static and determined by a single value. Others are dynamic; their evolution is controlled by an envelope — a normalized function consisting of linear and exponential segments — and a maximum size. Not all control parameters are totally independent; some occur only in certain combinations, and some are designed to reinforce others.

The control parameters give DIASS its flexibility and make it an instrument suitable for data sonification. On the other hand, the fact that the control parameters act at the level of a partial as well as at the level of a sound (or even at the level of a collection of sounds) significantly increases its computational complexity.

Table 1: Static (S) and dynamic (D) control parameters in DIASS.

Level	Description	Control parameter
Partial	Carrier (sine) wave	S: Starting time, duration, phase
		D: Amplitude, frequency
	AM (tremolo) wave	S: Wave type, phase
		D: Amplitude, frequency
	FM (vibrato) wave	S: Wave type, phase
		D: Amplitude, frequency
	Amplitude transients	S: Max size
		D: Shape
	Amplitude transient rate	S: Max rate
		D: Rate shape
	Frequency transients	S: Max size
		D: Shape
	Frequency transient rate	S: Max rate
		D: Rate shape
Sound	Timbre	D: Partial-to-sound relation
	Localization	D: Panning
	Reverberation	S: Duration, decay rate, mix
	Hall	S: Hall size, reflection coefficient

1.1.2 The Score

Input for DIASS consists of a raw score file detailing the controls. The raw score file is transformed into a score file for the instrument—a collection of "Instrument cards" (I-cards), one for each partial, which are fed to the instrument by M4C. The transformation is accomplished in a number of steps.

Among the controls are certain global operations ("macros"), which are defined at the level of a sound. In a first pass, these global controls are expanded into controls for the individual partials. The next step consists of the application of the loudness routines. These routines operate at the sound level and ensure that the sounds have the desired loudness. The final step consists of the application of the anticlip routines. For various reasons, historical as well as technical, sound samples are stored as 16-bit integers. The anticlip routines guarantee that none of the sample values produced by the instrument from the score file exceeds 16 bits. Because loudness and anticlip play a significant role in sonification, we discuss the issues in more detail.

Loudness. The perception of loudness is a subjective experience. Although the perceived loudness of a sound is related to the amplitudes of its constituent partials, the relation is nonlinear and depends on the frequencies of the partials. At the most elementary level, pure sinusoidal waves of low or high frequencies require a higher energy flow and therefore a larger amplitude to achieve the same loudness level as similar waves at mid-range frequencies. When waves of different frequencies are superimposed to form a sound, the situation becomes still more complicated. The sum of two tones of the same frequency produced by two identical instruments played simultaneously is not perceived as twice as loud as the tone produced by a single instrument.

An algorithm for data sonification must reflect these subjective experiences. For example, when we sonify two degrees of freedom, mapping one (x_1, say) to amplitude and the other (x_2, say) to frequency, then we should perceive equal loudness levels when x_1 has the same value, irrespective of the values of x_2. Also, when the variable x_1 increases or decreases, we should be able to perceive a proportional increase or decrease in the loudness level.

The loudness routines in DIASS incorporate the relevant results of psychoacoustic research [11] and give the user full control over the perceived loudness of a sound. They also scale each partial so each sample value fits in a 16-bit register (see Box 2).

Anticlip. When several sounds coexist and their waveforms are added, sample values may exceed 16 bits (overflow), even when the individual waveforms stay within the 16-bit limit. Overflow gives rise to "clipping"—a popping noise—when the sound file is played. The anticlip routines in DIASS check the score for potential overflow and rescale the sounds as necessary, while preserving the ratio of perceived loudness levels. Thus it is possible to produce an entire sound file in a single run from the score file, even when the sounds cover a wide dynamic range.

To appreciate the difficulty inherent in the scaling processes, consider the case of a sound cluster consisting of numerous complex sounds, all very loud and resulting in clipping, followed by a barely audible sound with only two or three partials. If the cluster's amplitude is brought down to fit the register capacity, and that of the soft tiny sound following it is scaled proportionally, the latter disappears under system noise. On the other hand, if only the loud cluster is scaled, the relationship between the two sound events is completely distorted. Many times in the past, individual sounds or groups of sounds were generated separately and then merged with the help of analog equipment or an additional digital mixer. The loudness and anticlip routines in DIASS deal with this problem by adjusting both loud and soft sounds, so their perceived loudness matches the desired relationship specified by the user, and no clipping occurs (see Box 3).

1.1.3 The Editor

Features like the loudness routines make DIASS a fine-tuned, flexible, and precise instrument suitable for data sonification. Of course, they require the specification of significant amounts of input data. The editor in DIASS is designed to facilitate this process. It comes in a "slow" and a "fast" version.

In the slow version, data are entered one at a time, either in response to questions from a menu or through a graphic user interface (GUI). The process gives the user the opportunity to build sounds step by step, experiment, and fine-tune the instrument. It is suitable for sound composition and designing prototype experiments in sonification. The fast version uses the same code, but reads the responses to the menu questions from a script. This version is used for sonification experiments.

1.1.4 Computing Requirements

The sound synthesis software embodied in DIASS is computationally intensive (see Box 4). The instrument proper, the engine that computes the samples, has been implemented in a workstation environment and on the IBM Scalable POWERparallel (SP) system. Parallelism is implemented at the sound level to minimize communication among the processors and enable all partials of a sound to access the same random number sequence. In parallel mode, at least four processors are used—one to distribute the tasks and supervise the entire run (the "master" processor), a second to mix the results (the "mixer"), and at least two "slave" nodes to compute the samples one sound at a time. Sounds are computed in their starting-time order, irrespective of their duration or complexity. (A smart load-balancing algorithm would take the duration of the various sounds and the number of their partials into account.)

Performance depends greatly on the complexity of the sounds—that is, on the number of partials per sound and the number of active controls for each partial. Typically, the time to generate a two-channel sound file for a 2'26" musical composition with 236 sounds and 4939 partials ranges from almost two hours on four processors to about 10 minutes on 34 processors on the SP. Figure 2 gives some indication of the speedups one observes in a multiprocessing environment. The three graphs correspond to three variants of the same 2'26" piece with different complexity. The time

T_p refers to a computation on $p+2$ processors (p "slaves"); all times are approximate, as they were extracted from data given by LoadLeveler, a not very sophisticated timing instrument for the SP. Speedup is measured relative to the performance on four processors (two compute nodes). One observes the typical linear speedup until saturation sets in. The more complex the piece (the more partials), the later saturation sets in.

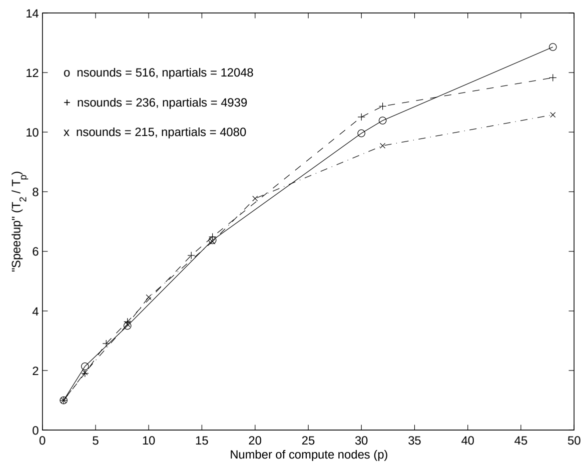


Figure 2: Timing results for DIASS on an IBM SP.

With a sampling rate of 44,100 samples per second and two-channel output, a sound file occupies 176 KBytes per second of sound, so the sound file for the 2'26" musical composition takes close to 25.8 MB of memory.

2 Data Sonification

Sonification is the faithful rendition of data in sounds. When the data come from scientific experiments— actual physical experiments or computational experiments— we speak of "scientific sonification." Scientific sonification is therefore the analog of scientific visualization, where we deal with aural instead of visual images. Because sounds can convey significant amounts of information, sonification has the potential to increase the bandwidth of the human/computer interface. Yet, its use in scientific computing has received limited attention. One reason is, of course, that our sense of vision seems much more dominant than our sense of hearing. Another important reason is the lack of a suitable instrument for scientific sonification. One of the goals of our project is to demonstrate that, with an instrument like DIASS, one can probe multidimensional datasets with surgical precision and uncover structures that may be hidden to the eye.

2.1 Past Experiments

An early experiment with scientific sonification was done by Yeung [29]. Seven chemical variables were matched with seven variables of sound: two with frequency, one each with loudness, decay, direction, duration, and rest (silence between sounds). His test subjects (professional chemists) were able to understand the different patterns of sound representations and correctly classify the chemicals with a 90% accuracy rate before and a 98% accuracy rate after training. His experiment showed that motivated expert users can easily adapt to complex auditory displays.

Recently, a successfull application of scientific sonification was reported in physics by Pereverzev et al. [15]. The authors were able to detect quantum oscillations between two weakly coupled reservoirs of superfluid ^{3}He using sound, where oscilloscope traces failed to reveal structure.

Several other experiments reported in the literature refer to situations where sounds are used in combination with visual images for data analysis. Bly [4] ran discriminant analysis experiments using sound and graphics to represent multivariate, time-varying, and logarithmic data. Mezrich et al. [14] used sound and dynamic graphics to represent multivariable time series data. The "Exvis" experiment at the University of Massachusetts at Lowell [20] expanded this work by assigning sonic attributes to visual icons. The importance of sound localization is recognized by ongoing work at NASA-Ames [27]. The evaluation of auditory display techniques is reported extensively at the annual conferences of ICAD, the International Conference on Auditory Display; see [12]. Sound as a component of the human/computer interface is discussed in [1].

Most of the attempts described above used MIDI-controlled synthesizer sounds, which have drastic limitations in the number and range of their control parameters. Bargar et al. [2] at the National Computational Science Alliance (NCSA) have developed a complex instrument with interactive capabilities, which includes the VSS sound server for the CAVE virtual-reality environment.

2.2 What We Have Done So Far

Much of our work so far has been focused on the development of DIASS [13, 10]. In addition, we have used DIASS for two preliminary experiments in scientific sonification, one in chemistry, the other in materials science.

The first experiment used data from Dr. Jeff Tilson, a computational chemist at ANL, who studied the binding of a carbon atom to a protonated thiophene molecule. The data represented the difference in the energy levels before and after the binding at $128 \times 128 \times 128$ mesh points of a regular computational grid in space. Because the data were static, we arbitrarily identified time with one of the spatial coordinates and sonified data in planes parallel to this axis. The time to traverse a plane over its full length was usually kept at 30 seconds. In a typical experiment, we assigned a sound to every other point in the vertical direction, distributing the frequencies regularly over a specified frequency range, and used the data in the horizontal direction to generate amplitude envelopes for each of the sounds. Thus, a sound would become louder or softer as the data increased or decreased, and the evolution of the loudness distribution within the ensemble of 64 sounds was an indicator of the distribution of the energy difference before and after the reaction in space. The sound parameters chosen for the representation of the data varied from one experiment to another.

The second experiment involved data from a numerical simulation in material science. The scientists were interested in patterns of motion of magnetic flux vortices through a superconducting medium. The medium was represented by 384×256 mesh points in a rectangular domain. As the vortices are driven across the domain, from left to right, by an external force, they repel each other but are attracted by regularly or randomly distributed defects in the material. In this experiment, frequency and frequency modulation (vibrato) were used to represent movement in the plane, and changes in loudness were connected to changes in the speed of a vortex. A traveling window of constant width was used to capture the motion of a number of vortices simultaneously.

These investigations are still ongoing, and the results have not been subjected to rigorous statistical evaluation. They have merely served to demonstrate the capabilities of DIASS and explore various mappings from the degrees of freedom in the data to the parameters controlling the sound synthesis process. Samples can be heard on the Web [25].

2.3 What We Have Found So Far

General conclusions are that (i) the sounds produced in each experiment conveyed information about the qualitative nature of the data, and (ii) DIASS is a flexible and sophisticated tool capable of rendering subtle variations in the data.

Changes in some control variables, like time, frequency, and amplitude, are immediately recognizable. Changes in the combination of partials in a sound, identifiable through its timbre, can be recognized with some practice. Some effects are enhanced by modifiers like reverberation, amplitude modulation (tremolo), and frequency modulation (vibrato). In some instances, a modifier may lump two, three, or more degrees of freedom together, like hall size, duration, and acoustic properties in the case of reverberation. Through the proper manipulation of reverberation, loudness, and spectrum, one can create the illusion of sounds being produced at arbitrary locations in a room, even with only two speakers.

Like the eye, the ear has a very high power of discrimination. Even a coarse grid, such as the temperate tuning used in Western music, includes about 100 identifiable discrete steps over the frequency range encompassed by a piano keyboard. Contemporary music, as well as some non-Western traditional music, successfully uses smaller increments of a quarter tone or less for a total of some 200 or more identifiable steps in the audible range. Equally discriminating power is available in the realm of timbre.

Sound is an obvious means to identify regularities in the time domain, both at the microlevel and on a larger scale, and to bring out transitions between random states and periodic happenings. Most auditory processes are based on the recognition of time patterns (periodic repetitions giving birth to pitch, amplitude or frequency modulation; spectral consistency creating stable timbres in a complex sound; etc.) and the ear is highly attuned to detect such regularities.

Most conceptual problems in scientific sonification are related to finding suitable mappings between the space of data and the space of sounds. Common sense points toward letting the two domains share the coordinates of physical space-time if these are relevant and translating other degrees of freedom in the data into separate sound parameters. On the other hand, it may be advantageous to experiment with alternative mappings. Sonification software must be sufficiently flexible that a user can pair different sets of parameters in the two domains.

Any mapping between data and sound parameters must allow for redundancies to enable the exploration of data at different levels of complexity. Similar to visualization software, sonification software must have utilities for zooming, modifying the audio palette, switching between visual and aural representation of parameters, defining time loops, slowing down or speeding up, etc.

Our experiments also showed that DIASS, at least in its present form, has its limitations. One limitation concerns the sheer volume of data in scientific sonification. While the composition of a musical piece (the original intent behind DIASS) typically entails the handling of a few thousands sounds, each with a dozen or so partials, the number of data points in the computational chemistry experiment ran into the millions, a difference of several orders of magnitude. By the same token, while a typical amplitude envelope for a partial or sound in a musical composition involves ten or even fewer segments, both experiments required envelopes with well over 100 such segments. Another difficulty encountered was the fact that both experiments required sounds to be accurately located in space. While panning is very effective in pinpointing the source on a horizontal line, suggesting the height of a sound is a major challenge. We hope that additions to the software as well as a contemplated eight-speaker system will help us get closer to a realistic three-dimensional representation of sounds. Finally, to become an effective tool for sonification, DIASS must operate in real time. All three concerns are being addressed in the new C++ version of DIASS currently under development.

3 Sound Visualization in a VR Environment

The notion of sound visualization may at first sight seem incongruous in the context of data sonification. However, as has been recognized by several researchers, the structure of a sound is difficult to detect without proper training, and any means of aiding the detection process will enhance the value of data sonification. Visualizing sounds is one of these means. In this project we are focusing on the visualization of sounds in the CAVE, a room-size virtual-reality (VR) environment [26], and on the ImmersaDesk, a two-dimensional version.

3.1 M4CAVE – A Visualization Tool

The software collectively known as M4CAVE takes a score file from the sound synthesis program DIASS and renders the sounds represented by the score as visual images in a CAVE or Immersadesk. The images are computed on the fly and are made to correspond exactly to the sounds one hears through a one-to-one mapping between control parameters and visual attributes. The code, which is written in C++, uses OpenGL for visualizing objects.

3.1.1 Graphical Representations

Currently, M4CAVE can represent sounds either as a collection of spheres (or cubes or polyhedra), as a cloud of confetti-like particles, or as a collection of planes.

The spheres representation is the most developed and incorporates more parameters of a sound into the visualization than either of the other. Sounds are visualized as stacks of spheres, each sphere corresponding to a partial in the sound. The position of a sphere along the vertical axis is determined by the frequency of the partial, and its size is proportional to the amplitude. A sound's position in the stereo field determines the placement of the spheres in the room. The visual objects rotate or pulse when tremolo or vibrato is applied, and their color varies when reverberation is present. An optional grid in the background shows the octaves divided into twelve equal increments. Figure 3 — taken from our Web site [25], where more samples can be found — shows a visualization of nine sounds with different numbers of partials.

The plane and cloud representations were designed more on the basis of artistic considerations. (Remember that the purpose of the visualization is to aid the perception of sounds.) The strength of the cloud representation is in showing tremolo and vibrato in the sound. The planes representation is unique in that it limits the visualization to only one partial (usually the fundamental) of each sound. The various representations can be combined, and the mappings chosen for each representation can be varied by means of a menu.

3.2 Preliminary Findings

We have used M4CAVE to explore various mappings from the sound domain to the visual domain. Besides the obvious short score files to test the implementation of these mappings, we have used score files generated with DIASS of various musical compositions, notably the "A.N.L.-folds" of Tipei [24]. A.N.L.-folds is an example of a *manifold composition* described in Box 1. Each member of A.N.L.-folds lasts exactly 2'26", comprises between 200 and 500 sounds of medium to great complexity. The picture of Fig. 3 was taken from a run of one of these A.N.L.-folds.

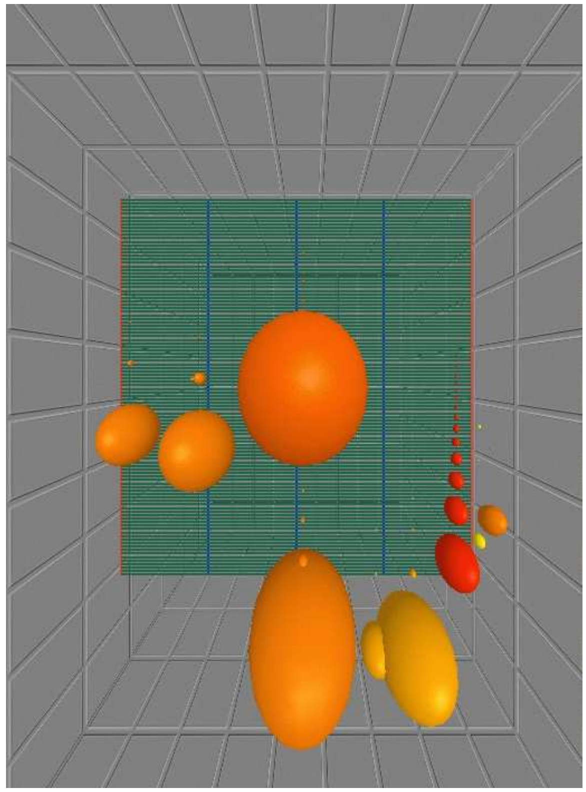


Figure 3: Visualization of nine sounds. (Picture taken from a CAVE simulator.)

The combination of visual images and sounds provides indeed an extremely powerfool tool for uncovering complicated structures. Sometimes, the sounds reveal features that are hidden to the eye; at other times, the visual images illuminate features that are not easily detectable in the sound. The two modes of perception reinforce each other, and both improve with practice.

4 Larger Issues

This project is unusual in several respects. It is somewhat speculative, in the sense that we don't have much experience with the use of sound in scientific computing. This is the main reason why the involvement of someone expert in the intricacies of the sound world is critical for its success. In our case, the expertise comes from the realm of music composition.

When do we declare "success"? Can we reasonably expect that sonification will evolve to the same level of usefulness as visualization for computational science? The answers to these questions depends on one's expectations. Ours is a visually oriented culture *par excellence*, and as a society we watch rather than listen. Contemporary musical culture is often reduced to entertainment genres that use a simple-minded vocabulary—no small impediment to discover the potential benefits of the world of sound. But with an awareness for unusual and unexpected sonorities, we may yet discover that we have not lost the ability to listen.

When we engage in this type of research, it is easy to get swept up by unreasonable expectations, looking for the "killer application." But the killer application is a phantom, not worth pursuing. What we can offer is a systematic investigation of the potential of a new tool. If it helps us understand some computational data sets a little better, or if it makes it a little easier to explore these data sets in more detail, we have good reason to claim success. If the project adds to our understanding of aural semiotics, we have even more reason to claim success. And if none of these successes materialize, we can still claim that the people involved, both scientists and musicians, gained by becoming more familiar with each other's work and ways of thinking. Such a rapprochement has, in fact, already occurred and led to a new "Discovery" course entitled *Music, Science, and Technology* at UIUC, where some of the issues presented here are being discussed in a formal educational context.

Acknowledgments

This work was partially supported by the Mathematical, Information, and Computational Sciences Division subprogram of the Office of Computational and Technology Research, U.S. Department of Energy, under Contract W-31-109-Eng-38.

References

[1] Baecker, R. M., J. Grudin, W. Buxton, and S. Greenberg, *Readings in Human-Computer Interaction: Toward the Year 2000*, second edition, Morgan Kaufmann Publ., Inc., San Francisco, 1995

[2] Bargar, R., I. Choi, S. Das, and C. Goudeseune, "Model-based interactive sound for an immersive virtual environment," *Proc. 1994 Int'l. Computer Music Conference* (Tokyo, Japan), pp. 471–477.

[3] Beauchamp, J., *Music 4C Introduction*, Computer Music Project, School of Music, University of Illinois at Urbana-Champaign, 1993. URL: http://cmp-rs.music.uiuc.edu/cmp/software/m4c.html

[4] Bly, S., *Sound and Computer Information Presentation*, PhD thesis, University of California – Davis, 1982 (unpublished)

[5] Fletcher, H. and W. A. Munson, "Loudness, its definition, measurement, and calculation," *J. Acoust. Soc. Am.* **5** (1933), 82

[6] Gropp, W., E. Lusk, and A. Skjellum, *Using MPI: Portable Parallel Programming with the Message-Passing Interface*, MIT Press, 1994. See also URL: http://www.mcs.anl.gov/mpi/index.html

[7] Hiller, L. and L. Isaacson, *Experimental Music*, McGraw-Hill, 1959; reprinted by Greenwood Press, 1983

[8] Hiller, L., *Computer Music Retrospective*, Compact disc WER 60128-50, WERGO Schallplatten GmbH, Mainz, Germany, 1989

[9] International Organization for Standardization (ISO), "Acoustics – Normal equal-loudness level contours," Publ. No. 226:1987

[10] Kaper, H. G., D. Ralley, J. M. Restrepo, and S. Tipei, "Additive synthesis with DIASS_M4C on Argonne National Laboratory's IBM POWERparallel System (SP)," *Proc. 1995 Int'l. Computer Music Conference* (Banff, Canada), pp. 351–352

[11] Kaper, H. G., D. Ralley, and S. Tipei, "Perceived equal loudness of complex tones: A software implementation for computer music composition," *Proc. 1996 Int'l. Conference in Music Perception and Cognition* (Montreal, Canada), pp. 127–132

[12] Kramer, G. (ed.), *Auditory Display: Sonification, Audification, and Auditory Interfaces*, Proc. ICAD '92, Addison-Wesley Publ. Co., 1994. For proceedings of later conferences, consult URL: http://www.santafe.edu/ icad

[13] Kriese, C. and S. Tipei, "A compositional approach to additive synthesis on supercomputers," *Proc. 1992 Int'l. Computer Music Conference* (San Jose, California), pp. 394–395

[14] Mezrich, J., S. Frysinger, and R. Slivjanovski, "Dynamic Representation of multivariate time series data," *J. Amer. Stat. Ass.* **79** (1984), 34–40

[15] Pereverzev, S. V., A. Loshak, S. Backhaus, J. C. Davis, and R. E. Packard, "Quantum oscillations between two weakly coupled reservoirs of superfluid ^{3}He," *Nature* **388** (1997), 449-451

[16] Roads, C., *The Computer Music Tutorial*, MIT Press, Cambridge, Mass., 1996

[17] Roederer, J. G., *The Physics and Psychophysics of Music*, 3rd edition. Springer-Verlag, 1995

[18] Rossing, T. D., *The Science of Sound*, Addison-Wesley Publ. Co., 1990

[19] Simoni, M. (ed.), *Proc. ICMC98, Int'l Computer Music Conference* (Ann Arbor, Michigan), October 1998; see also proceedings of earlier conferences

[20] Smith, S. and M. Williams, "The use of sound in an exploratory visualization experiment," CS Dept., U. Mass. at Lowell, Tech report R-89-002, 1989

[21] Stevens, S. S., "Neural events and the psychophysical law," *Science* **170** (1970), 1043

[22] Tipei, S., "The computer: A composer's collaborator," *Leonardo* **22**(2) 1989, 189–195

[23] Tipei, S., "Manifold compositions — A (super)computer-assisted composition experiment in progress," *Proc. 1989 Int'l. Computer Music Conference* (Columbus, Ohio), pp. 324–327

[24] Tipei, S., "A.N.L.-folds. mani 1943-0000; mani 1985r-2101; mani 1943r-0101; mani 1996m-1001; mani 1996t-2001" (1996). Report ANL/MCS-P679-0897, Mathematics and Computer Science Division, Argonne National Laboratory

[25] URL: http://mcs.anl.gov/appliedmath/Sonification/index.html

[26] URL: http://www.evl.uic.edu/pape/CAVE/prog/CAVEGuide.html

[27] Wenzel, E., S. Fisher, P. Stone, and S. Foster, "A system for three-dimensional acoustic 'visualization' in a virtual environment workstation," *Proc. Visualization '90: First IEEE Conf. on Visualization*, IEEE Computer Society Press, Washington, pp. 329–337

[28] Xenakis, I., *Formalized Music: Thought and Mathematics in Musical Composition*, revised edition, Pendragon Press, 1992

[29] Yeung, E., "Pattern recognition by audio representation of multivariate analytical data," *Analytical Chemistry* **52** (1980), 1120–1123

[30] Zwicker, E. and B. Scharf, "A model of loudness summation," *Psych. Rev.* **72** (1965), 3

[31] Zwicker, E. and E. Terhardt, "Analytical expressions for critical-band rate and critical bandwidth as a function of frequency," *J. Acoust. Soc. Am.* **68** (1980), 5

Hans G. Kaper is Sr. Mathematician at Argonne National Laboratory. After receiving his Ph.D. in mathematics from the University of Groningen (the Netherlands) in 1965, he held positions at the University of Groningen and Stanford University. In 1969 he joined the staff of Argonne. He was director of the Mathematics and Computer Science Division from 1987 to 1991. Kaper's professional interests are in applied mathematics, particularly mathematics of physical systems and scientific computing. He is a corresponding member of the Royal Netherlands Academy of Sciences. His main interest outside mathematics is classical music. He is chairman of "Arts at Argonne," concert impresario, and an accomplished pianist. He can be reached at kaper@mcs.anl.gov or http://www.mcs.anl.gov/~kaper/index.html.

Sever Tipei is professor of composition and music theory at the University of Illinois at Urbana-Champaign (UIUC), where he also manages the Computer Music Project of the UIUC Experimental Music Studios. He has a Diploma in piano from the Bucharest Conservatory in Romania and a DMA in composition from the University of Michigan. Tipei has been involved in computer music since 1973 and regards the composition of music both as an experimental and as a speculative endeavor. He can be reached at s-tipei@uiuc.edu or http://cmp-rs.music.uiuc.edu/people/tipei/index.html.

Elizabeth Wiebel is a participant in the Student Research Participation Program, which is sponsored by the Division of Educational Programs of Argonne National Laboratory. She is an undergraduate student at St. Norbert College in De Pere, Wisconsin, where she is pursuing a degree in mathematics and computer science. In addition, Wiebel studies and teaches piano through the St. Norbert College music department. She is currently spending a semester at Richmond College in London (England). She can be reached at wiebep@sncac.snc.edu, F300398@Richmond.ac.uk, or http://members.tripod.com/~LibbyW.

Box 1. Computer-Assisted Music Composition

The idea of using computers for music composition goes back to the 1950s, when Lejaren Hiller performed his experiments at the University of Illinois [7]. The premiere of his Quartet No. 4 for strings "Illiac Suite" [8] (May 1957) is generally regarded as the birth of computer music. Since then, computers have helped many composers to algorithmically synthesize new sounds and produce new pieces for acoustic as well as digital instruments. The proceedings of the annual conferences sponsored by the ICMA (International Computer Music Association) are good sources of references [19].

Why would a composer need computer assistance when composing? A quick answer is that, like in many other areas, routine operations can be relegated to the machine. A more sophisticated reason may be that the composer may rely on expert systems to write Bach-like chorales or imitate the mannerisms of Chopin or Rachmaninov. There are, however, more compelling reasons when composing is viewed as a speculative and experimental endeavor, rather than as an ability to manufacture pleasing sounds [22].

Music is basically a dynamic event evolving in a multidimensional space; as such, it can be formalized [28]. The composer controls the evolution by supplying a set of rules, and accepts the output as long as it is consistent with the logic of the program and the input data. If the set of rules allows for a certain degree of randomness, the output will be different every time a new "seed" is introduced. The same code and input data may thus produce an unlimited number of compositions, all belonging to the same "equivalence class" or *manifold composition* [23]. The members of a manifold composition are variants of the same piece; they share the same structure and are the result of the same process, but differ in the way specific events are arranged in time.

A nontraditional way of composing, the manifolds show how high-performace computing provides the composer with new means to try out compositional strategies or materials and hear the results in a reasonable amount of time.

Box 2. Loudness

Sound is transmitted through sound waves—periodic pressure variations that cause the eardrums to vibrate. But the perception of loudness has as much to do with the amount of energy that is carried by the sound wave as with the processing of this energy that takes place in the ear and the brain once the sound wave has hit the eardrums. The latter is a much more subjective part of the experience. The algorithms underlying the loudness routines of DIASS incorporate therefore formal definitions, as well as results of psychoacoustic research experiments. We summarize the most relevant elements of the algorithm, referring the reader to [17] or [18] for details.

The definition of (perceived) loudness begins with the consideration of the energy carried by the sound wave. The *intensity* I of a pure tone (sinusoidal sound) is expressed in terms of its average pressure variation Δp (measured in newton/m^2),

$$I = 20 \times \log_{10}(\Delta p / \Delta p_0).$$

Δp_0 is a reference value, usually identified with a traveling wave of 1,000 Hz at the threshold of hearing, $\Delta p_0 = 2 \times 10^{-5}$ newton/m^2. The unit of I is the decibel (dB).

Because of the way acoustical vibrations are processed in the cochlea (the internal ear), the sensation of loudness is strongly frequency dependent. For instance, while an intensity of 50 dB at 1,000 Hz is considered *piano*, the same intensity is barely audible at 60 Hz. In other words, to produce a given loudness sensation at low frequencies, a much higher intensity (energy flow) is needed than at 1,000 Hz. The intensity I is therefore not a good measure of loudness if different frequencies are involved.

In the 1930s, Fletcher and Munson [5] performed a series of loudness-matching experiments, from which they derived a set of *curves of equal loudness*. These are curves in the frequency (f) vs. intensity (I) plane; points on the same curve represent single continuously sounding pure tones that are perceived as being "equally loud." They are similar to those recommended by the International Organization for Standardization (ISO) [9] and are presented in Fig. 4. The curves show clearly that, in order to be perceived as equally loud, very low and very high frequencies require much higher intensities (energy) than frequencies in the middle range of the spectrum of audible sounds.

The (physical) *loudness level* L_p of a Fletcher-Munson curve is identified with the value of I at the reference frequency of 1,000 Hz. The unit of L_p is the phon. The Fletcher-Munson curves range from a loudness level of 0 to 120 phons over a

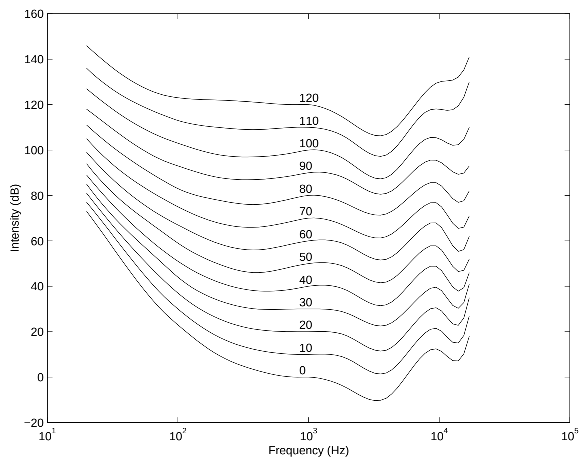


Figure 4: Curves of equal loudness (marked in phons) in the frequency vs. intensity plane.

frequency range from 25 to 16,000 Hz.

The loudness level L_p still does not measure loudness in an absolute manner: a tone whose L_p is twice as large does not sound twice as loud. Following Rossing [18], we define the (subjective) *loudness level* L_s in terms of L_p by the formula $L_s = 2^{(L_p-40)/10}$. The unit of L_s is a sone. To be effective, loudness scaling must be done on the basis of sones.

The loudness of a sound which is composed of several partials depends on how well the frequencies of the partials are separated. With each frequency f is associated a *critical band*, whose width Δf is approximely given by the expression [31]

$$\Delta f \approx 25 + 75\left(1 + 1.4(f/1000)^2\right)^{0.69}.$$

Intensities within a critical band are added, and the loudness of a critical band can again be read off from the Fletcher-Munson tables. If the frequencies of its constituent partials are spread over several critical bands, the loudness of a sound is computed in accordance with a formula due to Rossing [18],

$$L_s = L_{s,m} + 0.3 \sum_i L_{s,i}.$$

Here, $L_{s,m}$ is the loudness of the loudest critical band, and the sum extends over the remaining bands.

The loudness routines in DIASS use critical band information and a table derived from the Fletcher-Munson curves to create complex sounds of specified loudness.

Box 3. Loudness of Sound Clusters

The waveform of Fig. 5, which was produced with DIASS, illustrates the concept of equal loudness across the frequency spectrum and for different timbres. The waveform represents five sound clusters, each lasting 5.5 seconds (except the fourth, which lasts 5.7 seconds). The clusters, altough of widely different structure, have been designed to be perceived at the same loudness level (2^5 sones).

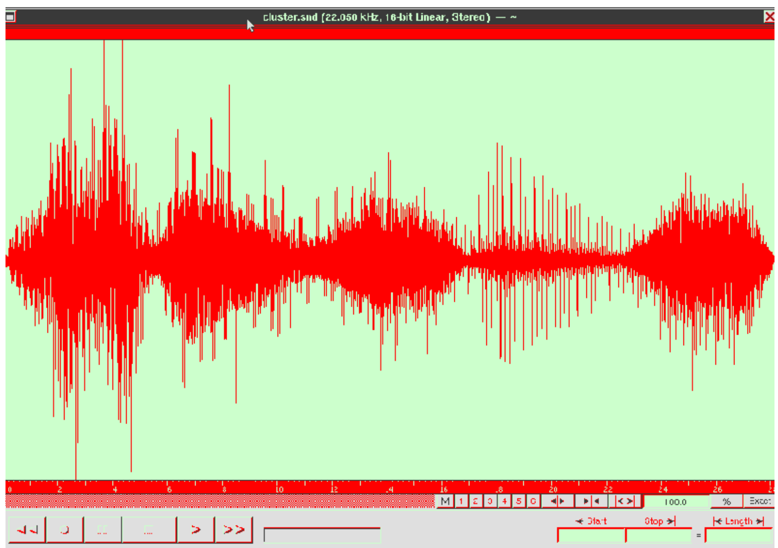
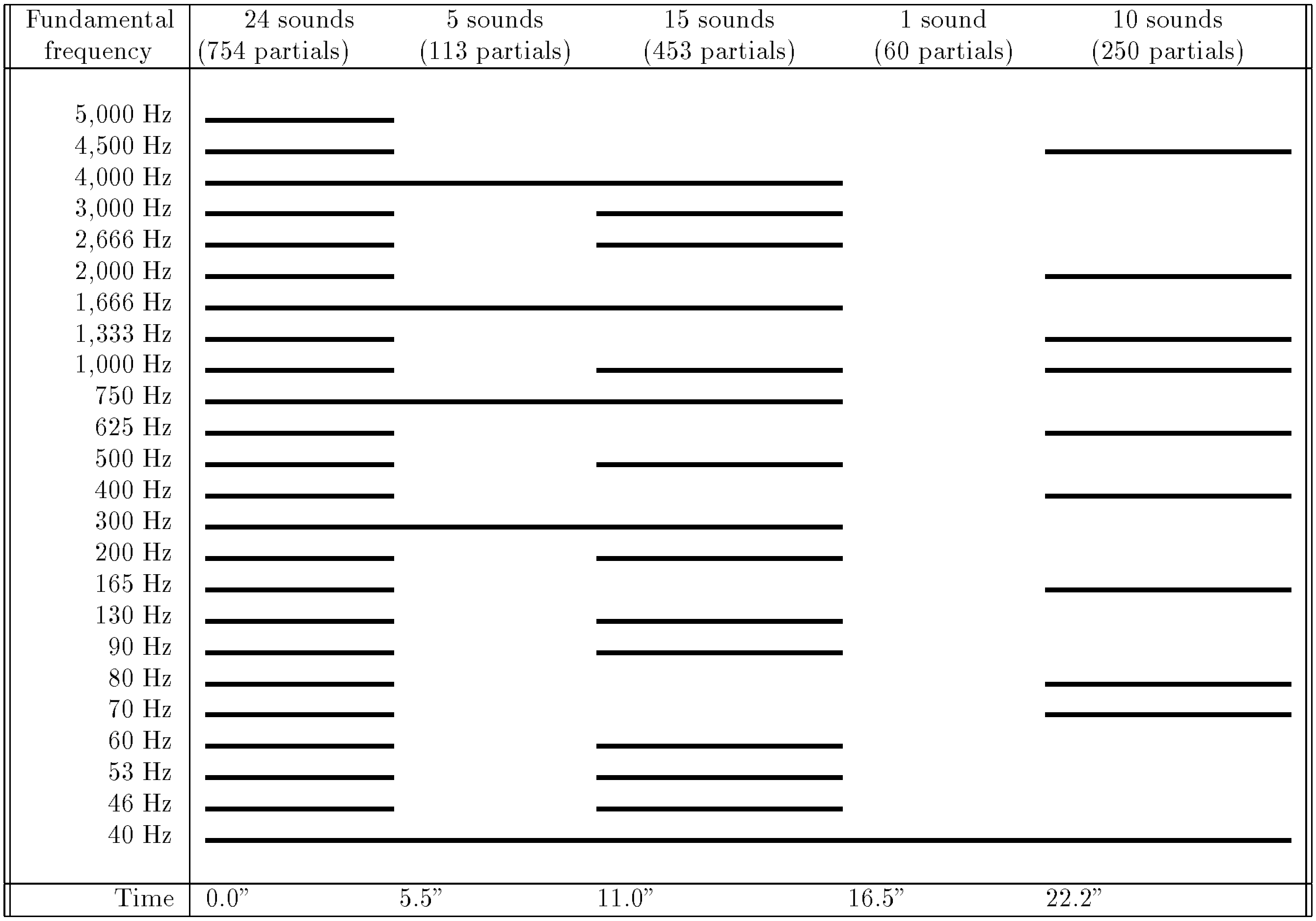


Figure 5: Waveform of five sound clusters of equal perceived loudness.

The distribution of the sounds within each cluster is represented schematically in the diagram of Table 2. The first sound cluster has 24 sounds. The fundamental frequencies of the sounds range from 40 to 5,000 Hz. Each sound is harmonically tuned; that is, it is made up of a fundamental and all its harmonics (partials whose frequencies are integer multiples of the fundamental frequency). The frequencies are limited to one-half of the sampling rate (Nyquist criterion); hence, the number of partials in this cluster is 754 (at a sampling rate of 22,050 Hz). The second sound cluster has 5 sounds, harmonically tuned, with fundamental frequencies ranging from 40 to 4,000 Hz; the number of partials is 113. The third, fourth, and fifth cluster have 15, 1, and 10 sounds, with 453, 60, and 250 partials, respectively. All partials are assigned the same amplitude, which presents the worst case scenario when one tries to obtain the same perceived loudness for all clusters.

Table 2: Distribution of fundamentals in the clusters of Figure 5.



Box 4. Computational Complexity

To give some idea of the computational complexity, consider the following simple scenario, where we wish to sonify time-varying data representing the values of two primary and several secondary observables measured over the course of an experiment. A natural choice is to map the primary observables onto loudness and frequency and to use amplitude and frequency modulation to monitor the secondary observables. The sample values of the sound wave S must be calculated from an expression of the form

$$S(t) = a(t) \sin (2\pi f(t) t + \phi) . \tag{1}$$

The frequency f represents three degrees of freedom: the carrier frequency f^C, and the amplitude a^{FM} and frequency f^{FM} of the modulating wave,

$$f(t) = f^C(t) + a^{FM}(t) \sin \left(2\pi f^{FM} t + \phi^{FM}\right) . \tag{2}$$

The carrier frequency is identified with a primary observable, each of the remaining two degrees of freedom can be identified with a secondary observable,

Similarly, the amplitude a is given by an expression of the form

$$a(t) = a^C(t) + a^{AM}(t) \sin \left(2\pi f^{AM} t + \phi^{AM}\right) . \tag{3}$$

We compute the carrier amplitude a^C from the observed loudness, which is identified with one of the (primary) observables, so its value is given. The amplitude a^{AM} and frequency f^{AM} of the modulation represent two more degrees of freedom, which can be identified with two other secondary observables. In total, we have therefore two primary and four secondary variables (not counting the phases, which we assume to be static).

The amplitude $a^C(t)$ must be computed such that $S(t)$ has the perceived loudness level $L_s(t)$,

$$L_s(S(t)) = L_s(t). \tag{4}$$

The loudness function L_s is a nonlinear function of the amplitude and frequency of the partial (sound). Its computation is done in the loudness routines of DIASS and involves a significant number of operations, including table lookups; see Box 2.

On the basis of these formulas we can obtain a rough estimate of the number of operations (additions, multiplications, function evaluations—sine, exponential, or logarithm, and table lookups) required for the computation of a single sample value. The contribution that is most difficult to estimate is the computation of the carrier

Table 2: Number of operations per partial per sample value.

Eq.	Adds	Mults	Fn Evals	Tbl Lkups
(1)	1	3	1	-
(2)	2	3	1	-
(3)	2	3	1	-
(4)	1	3	2	1
Total	6	12	5	1

amplitude from the loudness; the data in Table 2 represent the minimum number of operations. Ignoring phases, etc., we find a total of at least 24 operations. Hence, at the standard rate of 44,100 samples per second, one needs to perform more than 1.1 million operations per second.

The simultaneous sonification of more observables is obviously much more complicated; in fact, the complications grow exponentially. A careful estimate of the computational complexity requires an analysis of the anticlip routines, which is beyond the scope of the present article.