The Sound Manifesto

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ABSTRACT
Computing practice today depends on visual output to drive almost all user interaction. Other senses, such as audition, may be totally neglected, or used tangentially, or used in highly restricted specialized ways. We have excellent audio rendering through D-A conversion, but we lack rich general facilities for modeling and manipulating sound comparable in quality and flexibility to graphics. We need co-ordinated research in several disciplines to improve the use of sound as an interactive information channel.

Incremental and separate improvements in synthesis, analysis, speech processing, audiology, acoustics, music, etc. will not alone produce the radical progress that we seek in sonic practice. We also need to create a new central topic of study in digital audio research. The new topic will assimilate the contributions of different disciplines on a common foundation. The key central concept that we lack is sound as a general-purpose information channel. We must investigate the structure of this information channel, which is driven by the co-operative development of auditory perception and physical sound production. Particular audible encodings, such as speech and music, illuminate sonic information by example, but they are no more sufficient for a characterization than typography is sufficient for a characterization of visual information.

To develop this new conceptual topic of sonic information structure, we need to integrate insights from a number of different disciplines that deal with sound. In particular, we need to co-ordinate central and foundational studies of the representational models of sound with specific applications that illuminate the good and bad qualities of these models. Each natural or artificial process that generates informative sound, and each perceptual mechanism that derives information from sound, will teach us something about the right structure to attribute to the sound itself. The new Sound topic will combine the work of computer scientists with that of numerical mathematicians studying sonification, psychologists, linguists, bioacousticians, and musicians to illuminate the structure of sound from different angles. Each of these disciplines deals with the use of sound to carry a different sort of information, under different requirements and constraints. By combining their insights, we can learn to understand of the structure of sound in general.

Keywords: sound, user interface, audio, signal processing, acoustics

1. SOUND AS A TOPIC IN COMPUTER SCIENCE
A new topic of scholarly research is emerging in computer science—sound. Sound is normally studied from three different perspectives:

1. the production of sound by vibrating arrays of materials;
2. the propagation of sound through the air;
3. the perception of sound by the human ear and brain.

The production and propagation of sound are topics in physics, and the perception of sound is a topic in physiology and psychology. These three topics are often classified together as acoustics, but they are normally studied in isolation. Essentially, we have studied three critical things that happen to sound, without studying sound itself. Computer science provides the right perspective for a new study of sound as an inherently interesting object, connected intimately to, but not identified with, the mechanisms of production, propagation, and perception.

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The study of sound is not new, nor is the idea that it should be a topic in computer science. In 1993, Carla Scaletti and other forward-looking scholars held a Workshop on Sound-related Computation where they presented some visionary position papers announcing the mission of the Association for Computing Machinery’s new SIGSound group. With this Manifesto, we hope to bring the Sound topic some of the prominent attention that it deserves. This is the first version of the Manifesto, prepared for presentation at the conference on Critical Technologies for the Future of Computing, part of SPIE’s International Symposium on Optical Science and Technology in August 2000. We plan to expand and refine the Manifesto into a broad mission statement for the research community interested in sound.

1.1. Sound as an Information Channel

Sound is essentially an information channel. This is the hidden reason why particular phenomena in physics, physiology, and psychology are classified together as acoustics. The mechanisms for producing and for perceiving sound have developed co-operatively to allow information to be encoded in sound. Sound has an inherent information structure, determined by the combined capabilities of producers and perceivers, in which various encodings, such as music and speech, are realized. Study of the information structure of sound properly belongs at the center of the complex of acoustical topics, integrating insights from the study of production and of perception, and illuminating them both by explaining their purpose.

Computer science is accustomed to studying the structure of information, and algorithmic mechanisms for manipulating that structure. But the information structure of sound is substantially different from the channels that computer scientists have studied in the past. So, computer scientists can contribute a crucial perspective to the central topic of sound, while sound can stimulate fascinating new ideas in computer science.

1.2. Payoff from the Study of Sound

The systematic study of sound as an information channel will pay off in at least three important ways.

1. We will experience the scholarly joy of improving our understanding of an important phenomenon.

2. We will generate new techniques of analysis, modeling, and synthesis for existing applications of sound—speech recognition and synthesis, understanding of animal behavior, creation of music, etc.

3. We will exploit sound in human/computer interfaces, augmenting the rich interaction already provided by computer graphics, and making formerly visual materials accessible to visually handicapped people.

1.3. Current Specialized Sound Topics

Scientists already study a number of specialized topics concerning sound, including speech synthesis and analysis, linguistics, architectural acoustics, the behavior of musical instruments, identification of machines from their sounds, analysis of materials from their sounds, computer music, sonification. All of these specialties generate insights into the general structure of sound. But computer music and sonification are particularly important to my proposed systematic study of sound, because they study sound in general, with relatively little constraint from the intended application.

Just as typography exercises only a small and specially structured part of the space of visual symbols, speech exercises only a small and specially structured part of the space of sounds. On the other hand, the needs of musical composition and of sonification demand very broad and general palettes of sound, with the greatest possible flexibility in exploiting the natural structure of sound.

2. OVERLAPPING TIME SCALES IN SOUND

There are at least three different time scales for sound, which are perceived totally differently:

- In the sonic time scale, we sense the frequency of various sinusoidal components of a sound signal as pitch and color of a stationary sound. The sonic scale spans about $10^{-4}$ through $10^{-2}$ seconds. Sequences in the sonic time scale (e.g., sequences of samples) are not perceived as temporal sequences at all. The sonic time scale is analogous to the time scale in which the frequency of light is perceived as affecting color.
• In the transient time scale we perceive changes in the sinusoidal content of sound as attack, decay, and other transient effects. The transient time scale spans about $10^{-3}$ through $10^0$ seconds. Sequences in the transient time scale are also not perceived as temporal sequences. The analog in visual perception to the transient time scale in sound is not obvious, and perhaps there is none.

• In the event time scale, we perceive the relationship between transients explicitly as sequences in time. The event time scale spans about $10^{-2}$ through $10^2$ seconds. Perceptual time event sequences tend to combine events perceived by various senses, including acoustical and visual events.

To demonstrate the radically different perceptual structure of the three different auditory time scales, just play a familiar sound backwards. Time reversal has no impact on the mix of frequencies in the sonic time scale. Time-reversed transients change radically, and it is difficult or impossible to recognize the correspondence between the forward and reversed versions. Event sequences simply reverse themselves transparently. For another experiment, play a sequence of impulses at increasing frequencies. At 5 Hz, we clearly hear a sequence of clicks. At 1000 Hz we clearly hear a buzzing tone. We cannot locate the transition precisely. For some interesting portion of the 20–100 Hz region, we can hear a buzzing tone, and we can also resolve (but not count) individual clicks.

The temporal structure of sound is inherently more complicated than the temporal structure of vision, and may resemble that of haptic sensation. Visual perception also involves vibrational and event scales, but they are separated by orders of magnitude, and may be treated as independent dimensions. Due to this huge separation, animated graphics may be framed, and rendered as sequences of static images. No such framing can satisfy our perception of audible transients. Also, acoustical phase relationships are reported to the brain, while phase relationships in light are imperceptible, except to the extent that they reveal themselves by cancellation before they stimulate the retina.

3. DESCRIPTIVE MODELS OF SOUND

At the center of the study of sound lie descriptive models of sound. We need models that interpolate between the structure of sound producers and the structure of sound perceivers, capturing the information that is transferred from one to the other. We already have some excellent models of sound for certain purposes, but none of them satisfies the need for general-purpose models at the center.

1. Waveform models of sound as time sequences of amplitude values (also called time-domain models) are ideal for recording and rendering sound, and for certain mixing and signal-processing operations. They have little direct relation to either sound production or perception.

2. Fourier models of sound as sums of sinusoidal components at different frequencies (the frequency-domain models) capture some important qualities of perception at the cochlear (inner-ear) level for stationary sound. They represent transients and other time-development as cancellation properties of infinitely extended sinusoids, completely missing the structure of the perception of sonic change.

3. Time-frequency models of sound as sums of time- and amplitude-modulated sinusoidal components in the form

$$σ(t) = \sum_p A_p(t) \sin(Φ_p(t))$$

In this form, $t$ is time; each integer value of $p$ is the index of a component called a partial; $A_p$ and $Φ_p$ are the time-varying amplitude and phase for the $p$th partial; the derivative $dΦ_p/dt$ of phase is the time-varying frequency of the partial. There are infinitely many time-frequency descriptions of the same sound signal. With appropriate constraints on the $A_p$ and $Φ_p$, time-frequency models may be refined to match the cochlear level of perception very closely, and they represent a very important step in the refinement of sound models. Time-frequency models fail to capture higher levels of perceptual organization, particularly the natural relations between modulators of different partials. Perceptually very simple sounds have very complex time-frequency descriptions, and some perceptually natural transformations are very difficult.

4. Stochastic models of sound present some or all of a sound signal as the output of a stochastic process, instead of giving a numerically precise description. Since sound perception ignores a lot of the detailed differences between signals with the same statistical properties, stochastic models can improve the transparency and
representational efficiency of time-frequency models substantially. The simplest stochastic model presents sound as the sum of a deterministic component and white noise. More refined models employ filtered noise to match the spectral profile of a sound. Lemur\textsuperscript{14,15} associates noise with each partial, by introducing a stochastic component to each phase function $\Phi_p$. An ideal stochastic model should represent the perceptually significant qualities of the noise distributions of different partials, the distinction between random flutter in the phase and random jitter in the amplitude, and finally the correlation vs. independence between different random components.\textsuperscript{27}

5. Excitation/filter models of sound decompose sound into an excitation (which might be presented in a time-frequency description) and a filter with broad frequency response. Excitation/filter models make it easy to transform a sound by moving the excitation frequency, while holding the filter resonances constant. This corresponds fairly closely to the behavior of many instruments and the sung vowels, and sound perception seems to have adapted itself to such transformations.

6. Physical\textsuperscript{24,31} models of sound describe the configurations of vibrating materials that produce sound, usually with finite-element simulations. Physical models are very useful for understanding the behavior of real acoustical instruments,\textsuperscript{16} and take an important step in establishing constrained relationships between the time development of different sinusoidal partials. But they fail to capture the perceptual structure of those relationships. It is generally quite difficult to manipulate a finite-element model in order to accomplish an intuitively understood change in the sound that it generates. Manipulation of finite-element physical models is essentially a mathematical analog to the manufacture of acoustical musical instruments—not an intuitive task. Waveguide\textsuperscript{36} models organize finite-element systems and improve their computational complexity, but do little to improve intuitive transparency.

7. Modal\textsuperscript{1,5} models of sound are based on physical insights, but they organize sound producers according to their modes of vibration, rather than their geometrical material structure. Modal models can use matrices of interactions between different modes to constrain the structure of transients. This is an excellent idea, capturing just the sort of interpolation between production and perception that we need. The details require a lot more study.

4. SYNTHESIZING SOUND

Each descriptive model leads fairly naturally to synthesis techniques. While it is very challenging to implement these synthesis techniques efficiently enough for real-time processing, once the right models have been identified, concentrated research plus Moore’s law will probably provide us with sufficiently powerful real-time computations. So, we do not expect the synthesis problem to be a strategic determinant of the quality of sound processing.

5. ANALYZING SOUND

Every descriptive model provides a target for analysis, but the analysis problems are more deeply challenging than the synthesis problems. The analysis problems for time-frequency models of sound, and all of the more sophisticated models, are extremely underconstrained—each signal in the time domain has infinitely many descriptions, with many degrees of freedom. The core of the analysis problem is to find the right additional constraints.

The current state of the art in sound analysis is based on various types of windowed Fourier\textsuperscript{2} and wavelet\textsuperscript{17,37} transforms. These methods are all equivalent to running the input signal through a bank of filters tuned to various frequencies. All of these methods produce sequences of quadruples of time, frequency, amplitude and phase. All of them suffer from the tradeoff between time and frequency resolution inherent in linear transforms. Given the time-frequency to amplitude-phase function, several techniques link these points into frequency- and amplitude-modulated partials, by linking amplitude peaks at nearby times. McAulay/Quatieri analysis\textsuperscript{23} represents the state of the art in tracing spectral peaks. Marchand has improved the location of peaks using signal derivatives.\textsuperscript{21} Heterodyne analysis refines the two-step process by tracing each partial with a dynamically tunable filter. Filter-bank techniques can be refined to approximate cochlear stimulation very accurately, but we may wish to do better.

It is natural to suppose that, since auditory perception depends only on information derived from cochlear stimulation, analysis techniques based on wavelet transforms can provide satisfactory insight into the structure of sound. On the contrary, we expect that there are perceptually important structural properties of a sound that do
not affect our perception of that sound, but do affect its perceived relations to other sounds (e.g., after mixing, reverberation, distortion, or other naturally occurring transformations). And, there may be nonlinear analysis techniques that avoid the time-frequency resolution limits of linear transforms. Notice that current analysis methods are linear up to the time-frequency to amplitude-phase function, but the peak tracing methods are nonlinear. We should seek more profoundly nonlinear methods.

Keep in mind that the problem is not accuracy in reconstructing the original signal—perfect accuracy in reconstruction is achievable by filter-bank techniques. The problem is choosing one of the infinitely many solutions to a highly underconstrained analysis problem. We have no a priori definition of the “correct” analysis, but we might identify principles that lead to plausible consensus. The best way to seek these principles is probably through the idea of minimal description, but that is only a motivating idea, and does not prescribe the solution. Roughly, we would like the analysis corresponding to the simplest quasiphysical (probably modal) method for generating a sound. The analysis technique postulated here is almost sure to be ill conditioned (i.e., small changes in the signal lead to large changes in the analysis) at certain points, particularly at perceptually ambiguous points. For example, the distinction between a pair of beating partials and a single amplitude modulated partial will probably be ill conditioned.

6. AUDITORY USER INTERFACES

Graphical user interfaces (GUIs) have improved the quality of computer/human interaction immensely over teletype terminals. Even very primitive graphics is very effective in displaying the organization of different logical segments of interaction through the layout of windows on a display. Sound has the potential to carry similar amounts of information, but for the most part our computers just beep at us. Focused computer science research in sound will end this waste of a sense, with serious auditory user interfaces (AUIs).

Graphical presentations of information enjoy one large structural advantage over audible presentations: built-in browsability. A static two-dimensional display is a small database, and a viewer can browse it easily by moving her head, eyes, and attention. Cartesian coordinates provide a built-in indexing system. So the overwhelming majority of GUI interaction involves sequences of static displays. Nature provides a rich interactive structure even while the display is static.

Sound can only encode information temporally, so AUI software must provide all browsing and other interactive function algorithmically. This is both a curse, because it has delayed the invention of useful general-purpose AUIs, and a blessing, because it will force us to a deeper understanding of cognitive issues in the interactive presentation of information. Direct translation from visual to audible presentation is very valuable, particularly when the information involved is essentially geometric (e.g., the location of objects around a person). But the full value of the AUI requires methods to present abstract information directly to the ear, without translating from a visual presentation. Anyone who encounters a descriptive diagram during a session of recording for the blind experiences the insufficiency of translating information to sound through a graphical intermediary.

Projects, such as AsTeR and Emacspeak, have already demonstrated the feasibility of browsing in an AUI, and the importance of direct audible presentation with no graphical intermediary. But the AUI design space has only been sampled, and an immense amount of research remains before we sense its structure and boundaries.

7. LINEARITY VS. NONLINEARITY IN SOUND PRODUCTION

Sound as an information channel gets much of its structure from the interaction between linear and nonlinear phenomena. The near-linearity of many resonant devices (violin strings, air cavities) and of cochlear receptors makes modulated sinusoids the right structural building blocks for sound at the lowest level. Modest nonlinearities in sound production, transmission, and cochlear stimulation produce harmonic distortion, making harmonic and near-harmonic sequences of sinusoidal partials crucial organizational units at the next level of abstraction. Severe nonlinearities in acoustic generators (bow-string, air-vocal-cord interactions) tend to generate harmonic sequences of partials. They also produce the attack-transient patterns that are major carriers of sonic information. We have very little practical understanding of the natural constraints on the structure of attacks and other transients, and this lack is largely due to the essential nonlinearity of practical acoustic generators.

For the purposes of this manifesto, we will focus on linear vs. nonlinear theories of acoustic filters. The word “filter” suggests electronic equalizers, noise suppressors, and similar systems. But the same basic theories apply to
all resonating systems, and we are mainly interested in understanding natural acoustic generators through the theory of “filters.”

The theory of linear stationary (i.e., unclocked) filters\textsuperscript{32} is a beautiful area of essentially complete mathematics, based on the LaPlace transform.\textsuperscript{6} This theory provides a thorough understanding of the structure and behavior of all linear resonant devices, including characterizations of crucial qualities, such as stability. We have no similarly comprehensive and tractable theory for nonlinear filters, and there probably is none.

Most of our understanding of nonlinear filters consists of detailed treatments of special cases, such as the acoustic generators in musical instruments and vocal tracts. The only completely general theories of nonlinear filters that we can find are the Volterra and Wiener theories,\textsuperscript{33,34} which generalize the Taylor series approach to nonlinear functions. These theories fail to provide the precise sorts of characterizations that we depend on in linear LaPlace theory, and only a small minority of acoustical research projects employ them.

For research in sound, we need a theory that rivals the perfection of LaPlace theory, covering some useful subclass of all nonlinear resonant devices. We do not know precisely how to define this subclass, but it should include models of typical nonlinear phenomena that are important to sound, such as nonlinear acoustical generators and saturation effects in resonators. Volterra and Wiener theories deserve a careful study, in hopes that they can enlighten the search for a good definition of nonlinear filters for models of sound. While aiming for a similar degree of success with LaPlace theory, we should not insist on detailed analogies between the results. For example, we may find great value in nonlinear transforms that are not perfectly invertible. And, instead of characterizing stability, we may decide to restrict attention to a class of nonlinear filters that are all stable.

\section{8. BUILDING A RESEARCH COMMUNITY IN SOUND}

In order to stimulate new insights into the central topic of sound structure, we need to combine those who are already doing excellent research in connected areas into a community of sound research. One tool for forging such a community is a forum for publication. We have established such a forum for unrefereed reports in the Computing Research Repository (CoRR) supported by ACM, the Los Alamos e-Print archive, and the Networked Computer Science Technical Reference Library (NCSTRL). We encourage interested authors to contribute to the repository—description and instructions are found at http://xxx.lanl.gov/archive/cs/intro.html. We encourage editors to solicit authors of posted reports to submit to their journals, and we are looking for resources to found a new journal on sound.

We also encourage computer science research institutions and associations to recognize sound as a topic similar in scope to computer graphics. In particular, we encourage ACM to upgrade SIGSound to the full status of other special interest groups, with formal membership.

\section{9. AUDITORY SCENE ANALYSIS}

Auditory scene analysis\textsuperscript{7-9,12,13} takes a sound signal with one or more channels and computes a representation of the signal as the sum of sound sources of various types. Scene analysis is the ultimate sound modeling problem, analogous to model construction in computer vision, and it will force us to clarify the structure of general-purpose models of sound. Auditory scene analysis may try to mimic human perception, or it may aim at other representations. Scene analysis requires a method for modeling individual sound sources, and multichannel versions of the problem may require tracking those sources through space. Source identification and separation are key subproblems. In spite of recent increase in attention, practical results are very limited. Human listeners are far better at these tasks than any existing software.

In current implementations, the initial stage for extraction is usually one of the filter-bank analysis methods mentioned above, which yields a time-frequency analysis. From this representation auditory features are extracted and grouped into events by exploiting various cues, such as synchronized energy onset, continuity, common modulation, harmonic frequency relations, and so on.\textsuperscript{36,12} In addition to filter-bank methods, other techniques have been employed, such as correlograms and frequency warping.\textsuperscript{41}

The attribution of events to sources is accomplished by providing a set of rules, which essentially amount to assumptions about the types of sources allowed in the representation. Variations on the problem deal with constrained or predetermined sources that produce a range of known sounds, modeled sources, spatially localized sources, etc. Variant problems may also be characterized by specific audio contexts, such as music, speech, or a particular...
environment (traffic, industrial noise, etc.) We may also constrain the number of sources involved, or we may be interested in tracking only one or few important sources (e.g., signal-from-noise separation and voice extraction). The difficulty and scope of the problem solved and the success of the solution depend greatly on the source constraints.

It is clear that bottom-up signal processing alone will not yield a satisfying solution to the problem of auditory scene analysis. Human listeners are able to employ a variety of higher-level organizational information in order to facilitate source separation and identification. Expectations of source behavior are very important and are affected by previous exposure, aesthetics, training, etc. Furthermore, different auditory contexts require different amounts of organizational knowledge and specialized learning: almost anyone can separate speech from traffic noise, while following the lines played by different instruments of a string quartet generally requires ear training and possibly the study of composition. Additional complications arise from auditory illusions, such as our tendency to infer continuity and to hear sounds not present in the source.7,19

10. AESTHETICS AND EFFICACY

Auditory user interfaces and sonification of scientific data call for an unusual partnership between science/engineering and the arts, in which the arts contribute to goals in science and engineering. Partnerships between scientists and artists are nothing new. The great computer music institutes, such as IRCAM, CCRMA, ... have always fostered such collaboration. Most collaboration aims at using techniques from science and engineering to understand art, or to produce art. Problems in musical perception also provide motivation and paradigmatic examples that drive some scientific research in auditory perception. The reverse—application of artistic principles to science and engineering—is relatively rare. We hope to see an increase in the application of art to science and engineering, based on the intuitive connection between aesthetics and efficacy in presenting information.

Scientists and engineers often suffer from a sort of reverse snobbery when they present their ideas. Overly polished presentations are sometimes scorned on the assumption that time and effort were applied to style at the expense of substance. Foolish attempts at cute style can obscure information, as Tufte shows for graphical presentations.38–40 On the other hand, crudeness in presentation also obscures conceptual content. Many early efforts at visualization of scientific data through 3-dimensional animations were ineffective due to distractingly poor aesthetics in the production, and the need for some level of artistic production quality is well accepted in visualization. We expect a similar need for good aesthetic design even in the most pragmatically motivated projects in auditory presentation of information. Because of our poorer understanding of the structure of sound, compared to visual scenes, We expect an even greater need for help from artists who enjoy an intuitive ability to construct aesthetically pleasing sounds. There are already a few collaborations where musical artists contribute their aesthetic intuitions to scientific sound projects, and we hope to see more.

So far, we know of no general scientific study of the link between aesthetics and efficacy. That will be a very exciting new area of research for anyone who discovers how to approach the issue. There has already been some study of the value of musical sounds for encoding nonmusical information.3

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We refined the ideas in The Sound Manifesto through discussions and correspondence with Larry Fritts, Josef Jurek, Hans Kaper, Gregory Kramer, Karen Landahl, Sylvain Marchand, Daniel Margoliash, Peter Meijer, Partha Niyogi, Howard Nusbaum, Silvia Pfeiffer, Stephen Travis Pope, T. V. Raman, Davide Rocchesso, Howard Sandroff, Carla Scaletti, Fred Stafford, Robert Strandh, Sever Tipei, Damien Tromeur-Dervout, Paul Vickers. Ilia Bisnovatyi prepared most of the sound and graphics demonstrations.

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