5:22

GG17. Critical piezoelectricity in percolation. Didier Sornette (Physique de la Matière Condensée, CNRS UA190, Faculté des Sciences, Parc Valrose, 06034 Nice Cedex, France), Michel Lagier (Thomson-Sintra, DASM, zone des Bouillides, Sophia-Antipolis, 06561 Valbonne, France), and Stéphane Roux (Laboratoire d'Hydrodynamique et Mécanique physique, CNRS UA 857, Ecole Supérieure de Physique et Chimie Industrielle, 10 Rue Vauquelin, 75231 Paris Cedex 05, France)

The problem of the critical behavior of piezoelectricity in percolating media is introduced for the first time. Its relevance stems from the fact that piezoelectricity associates mechanical and electrical properties whose separated critical behavior have been extensively discussed in the literature. It should also yield useful information on the deformation modes of the structures. This problem is suggested by the recent development of low-density, compliant flexible piezoelectric ceramics used as acoustic transducers. The physical properties of these systems are just beginning to be explored experimentally, and their modelization is an open problem. The limiting case of strong heterogeneousness, where percolation ideas may apply and lead to simple and universal predictions, is addressed. The simplicity of the analysis comes from the clear separation of three scales, the microscopic grain size R, the percolation mesh size ξ, which diverges as the percolation threshold is approached, and the macroscopic size L of the system ($R \ll \xi \ll L$). The main result is that the direct piezoelectric effect exhibits a critical divergence as more and more tenuous compliant porous ceramics are considered.

5:34

GG18. Transducer nonlinearity theory. Y. F. Chou (Shanghai Acoustics Laboratory, Academia, Sinica, No. 456 Xiao Mu Qiao Road, Shanghai, People's Republic of China)

One normally regards a transducer as a linear device. However, as the working frequency of nonlinear parametric arrays becomes lower and the parametric arrays nearfield theory becomes popular, the nonlinear effect of a transducer cannot be tolerated and seriously affects the outcome of the parametric measurement. The present theory regards the transducer linearity and nonlinearity in the following ways: First, the so-called "linearity" is the fact that the transducer will induce some quantity of electricity when there is sound pressure applied on the surface of the transducer, and the transducer can be regarded as a static capacitor. As for the nonlinear process of the transducer, the transducer not only induces some quantity of electricity but also vibrates in the direction of the sound wave, so the transducer is regarded not as a static capacitor but as a time-varied capacitor controlled parametrically by the incident sound pressure. The following conclusions are reached: (1) The transducer nonlinearity is independent of the polarization voltage; (2) the transducer nonlinearity is not only related to the difference frequency but also to the primary frequency; (3) it is a function of the first-order sensitivity of the primary frequency; and (4) the transducer nonlinearity also depends on the transducer characteristics and its shape.

5:46

GG19. An estimation of the characteristics of an ultrasonic transducer by means of waveform analysis of its transient response. Daitaro Okuyama and Yoshizo Mori (Mining College, Akita University, Akita, 010 Japan)

A transient current is generated when a steplike voltage is applied to a piezoceramic transducer. It is shown that the frequency spectrum and the damped capacity obtained by an FFT analysis of the waveform have a close similarity to the frequency characteristics of the admittance and the damped capacity as measured by an ordinary method. To generate the steplike voltage with a 0.01-μs rise time, a mercury relay and a power source, whose internal impedance is less than 1 Ω, are used. The large capacity (about 500 μF) is connected parallel to the output terminal of the dc power source to obtain such a low impedance. The damped capacity is calculated from the data length, the value of the applied dc voltage, and the dc component of the frequency spectrum. The resonant characteristics of the transducer can be obtained from the results of FFT analysis on the transient current waveform. Results for various types of transducers, such as piezoceramic disks, cylinders, and honeycombs, etc., obtained by the present method show good agreement with those obtained under the ordinary techniques.

WEDNESDAY AFTERNOON, 16 NOVEMBER 1988 KOHALA/KONA ROOM, 2:00 TO 6:00 P.M.

Session HH. Musical Acoustics II: Digital Signal Processing in Music

Isao Nakamura, Cochairman
University of Electrocommunications
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James W. Beauchamp, Cochairman
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Chairman's Introduction—2:00

Invited Papers

2:05

HH1. Partial synchrony in musical sounds: Some recent results using time-variant spectral analysis. James W. Beauchamp and Robert C. Maher (School of Music and Department of Electrical and Computer Engineering, University of Illinois at Urbana-Champaign, Urbana, IL 61801)

Time-variant techniques have been used to analyze quasiharmonic musical signals in terms of

$$s(t) = \sum_{k=1}^{n} c_k(t)\cos\left(2\pi \int f_k(t)dt + \theta_k\right).$$

One type of synchronous behavior occurs when each harmonic amplitude obeys a definite monotonic relationship $c_k = F_k(c_1)$. This appears to be approximately true for nonvibrato reed-instrument tones, with the notable exception of their attack epochs. On the other hand, the functions relating the harmonic amplitudes of flute tones exhibit considerable hysteresis. Another type of synchrony has to do with harmonicity and whether the frequencies of the partials are harmonically locked together so that $f_k(t) = kf_1(t)$. Partials of vocal tones with vibrato appear to obey this relationship, but those of oboe tones do not. Examples of the analysis and synthesis of sounds with and without synchronous behavior will be given.

2:30

HH2. Digital signal processing aspects of digital musical instruments. Jun-ichi Fujimori and Hirokazu Kato (Insoft System Laboratories, Center for Musical Instrument and Software Development, Yamaha Corporation, 10-1 Nakazawa-cho, Hamamatsu, 430 Japan)

Digital signal processing techniques are being widely used in electronic musical instruments and their accessories. Today's digital musical instruments provide a greater variety of sounds and a more precise control than ever before. However, these instruments lack intuitive, direct controls making them difficult for most musicians to use. Two possible approaches are being considered to reduce their complexity. One is to utilize a powerful MPU for controlling the parameters of the digital signal processing, and the other is to find models of digital signal processing that are specific to music. In this paper, several techniques are surveyed, especially as applied to musical activities. In particular, some sound alteration methods are examined, leading toward a solution to the problem of the musician–machine interface.

2:55

HH3. Analysis and synthesis of sounds using waveform and source–filter models. Xavier Rodet (IRCAM 31 Rue Saint Merri, 75004 Paris, France)

Sound analysis and synthesis for music requires accurate analysis and high-quality synthesis of sounds. Some of the methods that were developed at the Institut de Recherches et de Communication Acoustique/Musique (IRCAM) for contemporary music research and production are described. The following analysis techniques have been implemented and used: FFT and phase-vocoder, wavelets, linear prediction algorithms (Levinson, global covariance, Burg, lattice covariance, lattice recursive), prony analysis, pitch detection, temporal envelope tracking (power tracking), detection of harmonic/noisy components, modal analysis (resonance modeling), formant-waveforms (FOF) coding, and formant trajectory coding. The phase-vocoder is used extensively for analysis and processing (filtering, cross-synthesis, time warping, etc.) of speech and music. Different LPC analysis algorithms have been studied and are used for precise estimation of spectral envelopes. These envelopes can be coded into (as well as recalculated from) formant parameters without loss of precision. Sounds and LPC residuals are coded into a *harmonic/noise* index for each of the frequency bands centered around harmonic partials of the signal, thus improving the quality of synthetic sounds. For sound processing, the following algorithms have been used: cascade filtering (LPC-vocoder), parallel filtering (in the program CHANT), frequency domain filtering (phase-vocoder), time-warping (phase-vocoder and LPC-vocoder), time-varying and arbitrary sample-rate conversion. Several synthesis methods have been developed, including LPC-derived synthesis filter, formant synthesis, formant-waveform (CHANT), parallel filtering, phase-vocoder, and additive synthesis. Synthesis-by-rule uses sophisticated object-oriented languages (PREFORM) for calculation, manipulation, and real-time control of synthesis parameters. For these studies, a real-time workstation is under development, based on SUN-3, Mercury array processor, and Sony PCM AD/DA.

3:20

HH4. String simulation by means of digital waveguides. Julius O. Smith (NeXT/CCRMA, 3475 Deer Creek Road, Palo Alto, CA 94304)

Research in digital sound synthesis and reverberation at CCRMA has led to a new approach, called "waveguide synthesis," in which acoustic traveling waves are explicitly simulated using digital delay lines. So far, the best success has been obtained for one-dimensional wave propagation, such as in woodwinds and strings. The ideal string is simulated using two delay lines, where one delay line carries the left-going traveling-wave component, and the other carries the right-going wave. Realistic strings are obtained by perturbing this lossless prototype using low-order digital filtering in cascade with the delay line. Because strings in musical instruments are close to lossless, it is highly efficient to simulate them as a weak perturbation of a lossless system. This paper reviews the fundamentals of string simulation using digital waveguides.