Digital Spectral Analysis of Audio Signals

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experiments illustrate the nature of variations in the ground-pressure distribution patterns due to aircraft accelerations and atmospheric effects. The phenomena of superboom generation during the transonic flight, atmospheric refraction, grazing incidence-wave impingement, and the interaction of shock waves with atmospheric turbulence and temperature anomalies are described, along with their significance regarding community response.

WEDNESDAY, 3 NOVEMBER 1965 REGENCY ROOM AT 10:00 A.M.

Session B. Signal Processing

THEODORE G. BIRDSALL, *Chairman*

Contributed Papers (12 min)

B1. *Sic Transit Sonitus.* F. V. HUNT, *Harvard University, Cambridge, Massachusetts 02138.*—Appropriate signal processing can make available a "count" or "lockup" signal when a moving source passes near a monitoring receiver. The rectified and filtered output of the sensor is sampled periodically and a "one" or a "zero" is delivered to a shift register according to whether the current sample is larger or smaller than the one before. For noise alone, ones and zeros accumulate in random sequence, but during a source transit the level at first increases monotonically and then decreases. After a transit, therefore, the $(2N+B)$-stage shift register will contain N ones in unbroken sequence, B indecisive digits that are ignored, and N zeros in another unbroken sequence. This unique $2N$-digit binary number can be recognized with a simple AND circuit, and the probability of it occurring by chance is only 2^{-2N}. A computer-aided study indicates that a detection probability of 90%, with a false-alarm rate of 10^{-6}, can be achieved for quiet slow sources whose signal-to-noise ratio at the point of closest approach is about −6 dB, and the detection range is almost never less for faster and noisier sources. [Work supported in part by the U. S. Office of Naval Research.]

B2. Detection Theory: Uncertain Signal-Arrival Times. L. W. NOLTE, *Electrical Engineering Department, The University of Michigan, Ann Arbor, 48105.*—The design, in block-diagram form, and performance of the optimum adaptive receiver is presented for a signal with synchronous Poisson arrival times. This receiver was evaluated experimentally on a digital computer for various values of duty factor, signal-to-noise ratio, and time. The receiver performance, presented in terms of the receiver operating characteristic (ROC), shows the effect of uncertain arrival times on detectability.

B3. Amplitude Learning in the Sequential-Clipper Crosscorrelator Detection Receiver. CHRISTOPHER V. KIMBALL (nonmember), *Electrical Engineering Department, The University of Michigan, Ann Arbor, 48105.*—A comparison is made between an adaptive sequential-clipper crosscorrelator and the adaptive nonclipping sequential receiver for the case of a signal known except for amplitude; i.e., amplitude is specified by a probability distribution. Both receivers update distributions on the signal amplitude during the observation process to learn the transmitted signal amplitude and, consequently, improve receiver performance. This study compares the amplitude learning of the two receivers during one step of the sequential procedure. Assuming the same initial information, a fixed small-amplitude signal is applied to the input of each of the receivers and the distributions after observation are studied. From these distributions, the amplitude learning of the sequential-clipper crosscorrelator receiver is compared with that of the amplitude-utilizing receiver to provide a measure of the efficiency of the clipper crosscorrelator receiver in learning the signal amplitude. The measure obtained is analogous to the well-known detection efficiency of $2/\pi$. [Work supported by the U. S. Office of Naval Research Acoustics Programs (Code 468).]

B4. Digital Spectral Analysis of Audio Signals. K. L. DOTY (nonmember), *Systems Development Division Laboratory, International Business Machines Corporation, Poughkeepsie, New York.*—Digital spectral analysis offers certain advantages over analog techniques: (1) the equivalent bandwidth of the analysis can be varied easily and (2) complicated calibration procedures are not required. A real-time analog-to-digital magnetic-tape recording system, designed as part of future instrumentation that will record digitized noise or speech signals and calculate their spectra with special purpose computer, records 31 250 digitized analog time samples per second in a 9-bit code. Currently, an IBM-7094 data-processing system is programmed to perform trapezoidal integration of the Fourier integral to obtain the spectrum of the digital recordings. The Fourier transform of a 16-msec square pulse differed from the calculated values by less than 4% of the actual amplitudes in the frequency range 0–5000 cps. A comparison was made between the digital spectra obtained for several speech sounds and those obtained with 50-cps constant-bandwidth analog filters. Within expected limits, the agreement was good. The time required for the IBM-7094 computer analysis and plotting program to process a single pitch period of the vowel "a," sustained at 125 cps, is 15 sec.

B5. New Method for the Rapid Determination of Speech Frequencies. GORDON R. PARTRIDGE, *General Radio Company, West Concord, Massachusetts.*—A new approach has been taken to the problem of rapid determination of the nature of sounds. The times between successive cycles of speech or other signals are measured, and the *reciprocals* of these time intervals are displayed on a storage oscilloscope. This results in an almost instantaneous presentation of frequency as a function of time, the readout of information lagging the acquisition of data by just a single cycle (a cycle is two successive zero crossings for purposes of this paper). A separate portion of the storage screen displays amplitude versus time. The patterns produced by the instrument look very similar to those of "visible speech" generated by other methods such as comb filters. Ultrasonic as well as audible frequencies are covered, up to 100 kHz. In these higher frequencies, the instrument has been used for the examination of bat and porpoise sounds.