PROGRAM Thirty-fourth Convention...Exhibition of Professional Products...

AUDIO ENGINEERING SOCIETY

April 29 through May 2, 1968

HOLLYWOOD ROOSEVELT HOTEL
7000 HOLLYWOOD BOULEVARD
HOLLYWOOD, CALIFORNIA

1:30 P.M. — THURSDAY, MAY 2, 1968

MUSIC AND SPEECH

CHAIRMAN: M. V. MATHEWS, Bell Telephone Laboratories Incorporated, Murray Hill, New Jersey

SOME TIME DEPENDENT CHARACTERISTICS OF CORNET, OBOE, AND FLUTE TONES—James W. Beauchamp, University of Illinois, Urbana, Illinois

A method previously described was used to extract the harmonic amplitudes and relative phases as functions of time for a large number of musical instrument tones. Motion pictures of harmonic line spectra changing in slow motion are presented for several tones to illustrate their time dependence. Some features of the attack characteristics of cornet, oboe, and flute tones, the importance of relative phases for oboe vibrato, and the separation of cornet spectra into effects produced by an intensity-dependent excitation function and frequency-dependent filter will be discussed.

SOUND PRODUCTION IN THE WIND INSTRUMENTS—John Backus, University of Southern California, Los Angeles, California

The wind instruments are self-excited acoustical oscillators which utilize the regenerative mechanism of the instrument reed to maintain complex oscillations in an approximately cylindrical or conical air column. The harmonic structure of this complex vibration and hence the radiated sound will depend on how much the resonance frequencies of the air column deviate from the frequencies of the harmonics of the air column vibration.

AN ALGORITHM FOR SEGMENTATION OF CONNECTED SPEECH—D. R. Reddy and P. J. Vicens, Stanford University, Stanford, California

Smoothing and differencing operations on the digitized acoustic waveform generate parameters used to determine whether the characteristics of the sound are changing or similar; parts that possess similar parameters are grouped together to form sustained segments and parts that possess changing parameters form transistional segments, resulting in the segmentation of connected speech into parts approximately corresponding to phonemes.

PREPROCESSING OF SPEECH FOR ADDED INTELLIGIBILITY IN HIGH AMBIENT NOISE—Ian B. Thomas and Russell J. Niederjohn, University of Massachusetts, Amherst, Massachusetts

Experiments are described in which the intelligibilities of preprocessed and normal speech are compared when a high level of ambient noise exists at the listener's ears. Preprocessing involves removal of the first formant of the speech prior to infinite amplitude clipping. The resulting clipped speech is highly intelligible;

under high noise conditions its intelligibility is considerably greater than that of normal speech at a comparable average sound level.

A SYMBOLIC APPROACH TO MUSICAL COMPOSITION—L. Knopoff, Institute of Ethnomusicology, University of California, Los Angeles, California

An investigation has been made into the possibility of generative communicable music by constructions using a symbolic form of composition. The number of symbols is small; in the computer-performed pieces to be heard as examples it is three. Each symbol controls an identical group of sounds each time it occurs. The generation of a musical style will be discussed and exhibited.

THE RELEVANCE OF MUSICAL THEORY TO MUSICAL DATA-PROCESSING—Michael Kassler, Washington, D. C.

Those who would process musical data non-haphazardly must process in accordance with some musical theory. The presence of automatic computers and the potential presence of specifically musical input/output devices have attracted attention to musical theories that have been explicitly expressed and are readily susceptible of restatement in algorithmic form. This paper will contrast the current state of computer processing of musical data with a more advanced state that would result from further development of explication of existing, albeit unexplicated, musical theories.

THE STANFORD SYSTEM FOR ON-LINE GENERATION OF SOUND BY COMPUTER—David Poole, John Chowning and Leland Smith, Stanford University, Palo Alto, California

As part of a project in computer-generated music at Stanford University, a flexible system has been developed for producing and manipulating sound with a digital computer. The implementation utilizes a general purpose time-sharing system, and provides visual and auditory interaction with the user in the generation of arbitrarily complex sounds. Initial work has been with artificial reverberation and simulated movement of sound sources.

MUSICAL SIGNAL PROCESSING FOR FUN AND PROFIT—Robert A. Moog, R. A. Moog Co., Trumansburg, New York

Dynamic variations of the timbres of musical sounds can be achieved electronically in many ways. This paper discusses circuitry that preserves the pitch and timing of monophonic musical sounds, but transforms the waveform and envelope. Examples will be played.