Two distinct tree search strategies (the Viterbi algorithm and the stack algorithm) were simulated in assembly language on a large digital computer. For the case investigated they were found to be improved by several modifications:

i) extra code digits supplied at the beginning of transmission to reduce a start-up problem;
ii) separate encoding of source outputs of magnitude greater than about 3.5 or 4 source standard deviations;
iii) a technique to prevent the stack algorithm from stalling in certain sections of data.

With these modifications, both strategies were made to simulate encoding with a signal-to-distortion ratio about 1 dB below the limit set by the rate-distortion function. This is a rate-distortion performance better than that of any previously known instrumentable scheme for this case. Unfortunately, this performance requires a tremendous amount of computation. For this performance each strategy was found to require on the order of 10^5 machine cycles per source output encoded.

Alton L. Gilbert, "Linear minimum-mean-squared-error estimation of exponentially correlated processes with application in interframe correlated video," Sc.D., Dep. Elec. Eng., New Mexico State Univ., Las Cruces, N. Mex., April 1973. Adviser: Frank F. Carden.

Exponentially correlated processes are among the most commonly used models of random processes for a wide variety of physical problems, including video processes. The examination of estimation techniques for sampled exponentially correlated processes in various degrading environments is therefore of considerable practical interest. The purpose of this study is to develop linear processors for the estimation of sampled exponentially correlated processes in noise which is uncorrelated for samples separated by integral multiples of the frame interval and in noise with a first-order Butterworth power density spectrum. The fidelity criterion used throughout is minimum mean-squared error.

Analysis is performed on the basis of a statistical model defined by its first two moments to develop an optimal linear minimum-mean-squared-error (LMMSE) estimator using three samples from two distinct data sets, yielding an interpolative and a recursive estimator for noise which is uncorrelated for samples separated by integral multiples of the frame interval. Theoretical curves are plotted showing the improvement realizable as a function of the correlation between samples and the signal-to-noise ratio (SNR). Discussion of the sensitivity of the models to parameter changes and verification of model validity for known parameter values demonstrate the usefulness and practicality of the models developed.

Extension of the results to a general $(2j + 1)$-stage estimator is included. Matrix results are derived to find the inverse of the covariance matrix necessary to explicitly find the general model. The general model is developed for both uncorrelated and exponentially correlated (first-order Butterworth) noise for samples separated by integral multiples of the frame interval, and improvements plotted for each as a function of correlation between successive samples and SNR. Improvement bounds are numerically determined and performance as compared to the bounds is shown for various numbers of stages. Derivation of the improvement bound for recursive estimation is shown to be the same as the general interpolative model.

Application of these results to interframe correlated video is the motivation throughout. A method for real-time application of the models developed is discussed.

Stephen C. Greenhouse, "On-line recognition of speakers by machine," Ph.D., Dep. Elec. Eng., Catholic Univ. America, Washington, D.C., May 1973. Adviser: Dr. Tālivaldis I. Šmits.

An investigation into the feasibility of the on-line recognition of speakers by digital computer is conducted. A speaker recognizer that decides, from a small sample of his speech, which member of the set of four reference (known) speakers is the test (unknown) speaker, is designed, implemented with software, and tested with actual data. Experimental results are compared with lower bounds on the probability of correct identification.

Single vowel phonemes, "taken out of context," are used as the sample utterances or patterns upon which the recognition scheme is based. Walsh functions are used as the basis of the transformation between measurement space and feature space for fast processing. The speaker feature vector is assumed to be an N-dimensional Gaussian random vector. Due to the fast processing requirement, certain assumptions about the covariance matrix are made, resulting in the maximum cross correlation or matched filter and estimator-correlator recognizer configurations. Results due to these assumptions are compared with the general Gaussian recognizer realization, in which no assumptions about the covariance matrix are made.

Estimator-correlator results are markedly superior to those of the matched filter configuration and even surpass the general recognizer above $N = 4$, indicating errors in estimating the covariance terms or in the Gaussian assumption itself above this point. Averaged all four phonemes, best performance (maximum percentage of correct identifications) of 81.2 percent occurs at $N = 36$ for this realization, which is as expected since this is approximately the dimensionality of the measurement space. Using all phonemes to make one decision, best performance of 83.3 percent occurs earlier at $N = 12$.

William R. Hahn, "Optimum estimation of a delay vector caused by a random field propagating across an array of noisy sensors," Ph.D., Dep. Elec. Eng., Univ. Maryland, College Park, Md., 1972. Adviser: Steven A. Tretter.

The problem of optimally processing multiple sensor data to determine the set of time delays generated by the propagation across an array of the wavefronts from a distant wide-band Gaussian noise source is investigated. It is assumed that the amplitude gradient across the array of the noise field is negligible, that the array outputs are corrupted by additive wide-band Gaussian independent sensor noises, and that the observation time is long. The Fisher information matrix is determined, and then used to show that the maximum likelihood estimate is asymptotically efficient (as theory dictates it should be). It is also shown that filtered correlator systems can provide asymptotically efficient estimates. Finally, the effects of suboptimal filtering of the inputs to a correlator system is investigated for the case when the signal and additive noise spectra are all band limited and have constant slopes of 0, -3, or -6 dB/octave.