PURDUE UNIVERSITY
SCHOOL OF ELECTRICAL ENGINEERING
ELECTRONIC SYSTEMS RESEARCH LABORATORY

SEMI-ANNUAL REPORT OF RESEARCH

PERFORMED UNDER GRANT NsG—553

July 1, 1967 through December 31, 1967

LAFAYETTE, INDIANA

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Foreword

This report summarizes work carried out at the Electronic Systems Research Laboratory of Purdue University under NASA Grant NsG-553 during the period July 1, 1967 through December 31, 1967.

In keeping with NASA's policy for administration of research grants, the report has been kept as concise as possible and, when appropriate, reference has been made to interim reports, internal memoranda, and technical papers resulting from research carried out under this grant.

The format of the report consists of a listing of technical papers and internal memoranda which have been submitted to NASA during the period covered by the report, abstracts of interim reports submitted during the reporting period, and appropriate extracts from the current Purdue University, School of Electrical Engineering, Semi-Annual Research Summary.

John C. Lindenlaub
Principal Investigator

I. Technical Papers and Internal Memoranda

Three technical papers summarizing research results obtained as a result of the grant were prepared during the reporting period. "Optimum Quantization," by A. J. Kurtenbach and P. A. Wintz was presented at the 1967 IEEE Eastcon Conference held in Washington, D.C. October 16-18, 1967. This paper was also published in the IEEE Transactions on Aerospace and Electronic Systems, Vol. AES-3, No. 6, November, 1967.

"Synchronization and Bit Timing," by P. A. Wintz was presented at the 1967 NEC Convention in Chicago, October 23-25, 1967 and is published in the proceedings of that conference.

A paper co-authored by D. R. Anderson and P. A. Wintz entitled "Analysis of a Multiple Access Satellite Communication System" has been submitted for publication to the IEEE Transactions on Communication Technology.

II. Abstracts of Interim Reports

An interim report entitled* "Digital Communication Systems Subject to Frequency Selective Fading" by J. C. Lindenlaub and C. C. Bailey has been completed and copies of the report mailed to NASA Headquarters on December 13, 1967. An abstract of this report is given below:

This research is concerned with the effects of frequency selective fading on binary digital communication systems and with a method of alleviating some of these effects. The frequency selective fading is assumed to be describable in terms of a filter whose transfer function is a sample function from a complex Gaussian random process. Matched filter detection at the receiver is assumed to be employed. Although the analytical methods used in this report are general, DPSK signaling is assumed for all the quantitative results.

In order to determine the probability of error for this system, a technique due to Bello and Nelin is used. This is applied to three cases:

1. Square pulse signaling and Gaussian-function-shaped channel frequency correlation function.
2. Square pulse signaling and $\sin(x)/x$ - shaped channel frequency correlation function.
3. Raised cosine signaling and $\sin(x)/x$ - shaped channel frequency correlation function.

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* TR-EE67-17, School of Electrical Engineering, Purdue University, Lafayette, Indiana, November, 1967.

The results of these three cases are compared and it is shown that significant differences in error probability result from using the two different signals with the same channel while very little difference in performance results from using the two difference channels with the same signal.

For the three cases mentioned above, an approximate error probability expression developed by Sunde is compared to the exact results. It is shown that this approximation does not provide a good estimate of the error probability when the frequency selective fading is of importance.

Next, some experimental results from computer simulations of communication systems subject to frequency selective fading are given. The first of these is an investigation of the relative effects of the amplitude and phase distortion introduced in such a system. It is shown that the phase distortion component has a deleterious effect on the system's performance while the amplitude distortion has a slight beneficial effect. These results provide the motivation for the development of an adaptive scheme for counteracting the effects of frequency selective fading. This is done by measuring the residual delay and delay distortion introduced by the channel fading with pilot tones and then using phase correction networks at the receiver to compensate for the measured phase characteristic. It is shown that this technique can result in error rate improvement factors of twenty or data rate increases by a factor of two to three. It is also shown that the system which corrects for both delay and linear delay distortion provides very little improvement over the system which corrects for delay only.

III. Research Summaries

The following sections present brief summaries of current research projects being carried out under the sponsorship of this grant.

A. An Investigation of Techniques for Analog Communication Over Selective Fading Channels

J. C. Lindenlaub

D. P. Murray

The purpose of this research has been to find feasible techniques for improving the performance of communication systems operating over slowly time-varying, frequency-selective fading channels. To judge the effectiveness of relatively simple demodulating systems, optimal signal processing techniques for communicating over known channels were derived, and their performance analyzed. It was found that there is an intrinsic loss in performance associated with selective fading, relative to non-fading channels. A direct relationship between the performance and complexity of the required optimal processing has also been established.

Several simpler, suboptimal demodulators using transversal equalizers and post-equalization filters were studied. Their performance was found to be near-optimal under certain fading conditions. The range of conditions for near-optimal operation widens as increasingly complex systems are considered.

For the case of unknown channels a channel measurement is performed using either the information-bearing signal alone, or a transmitted reference signal multiplexed with the message. In both cases optimal channel estimators were found, and in the latter case the reference signal was optimized as well. The non-reference technique requires such extensive data processing that it is impractical to implement, but it does lead to a bound on the performance of any system subject to fixed transmitter power. On the other hand, the optimal transmitted reference system is simply mechanized with analog components, and it has been shown that the measurement requires a negligible amount of transmitter power if the fading rate is slow compared to the system bandwidth.

The overall performance of combined channel and message estimating systems has been calculated for single-notch selective-fading channels, including a study of optimal transmitter power division. Both optimal and suboptimal techniques were investigated, and the tradeoff between complexity and performance established. It appears that the best balance between complexity and performance is achieved by a frequency-division multiplexed, transmitted reference system, using a minimum-mean-square transversal filter as a demodulator, and a binary PN sequence as a reference signal. This system is realizable with analog components, and its overall performance closely approaches the afore-mentioned performance bound under a wide variety of channel conditions.

B. Digital Communication Over Frequency Selective Channels

J. C. Lindenlaub

C. C. Bailey

An adaptive receiver scheme for combating effects of frequency selective fading in binary digital communication systems was described in a previous research summary (1). The project to evaluate the performance of this scheme as well as to evaluate the effect of frequency selective fading on conventional communication systems has been completed. A detailed description of the results of this project can be found in reference 2.

Throughout this work, the frequency selective fading was assumed to be describable in terms of a filter whose transfer function is a sample function from a complex Gaussian random process. Matched filter detection at the receiver was assumed to be employed. Although the analytical methods used in this project were general, DPSK signaling was assumed for all the quantitative results.

In order to determine the probability of error for the conventional nonadaptive system, a technique due to Bello and Nelin was used. This was applied to three cases:

1. Square pulse signaling and Gaussian-function-shaped channel frequency correlation function.
2. Square pulse signaling and sin(x)/x - shaped channel frequency correlation function.
3. Raised cosine signaling and sin(x)/x - shaped channel frequency correlation function.

The results of these three cases were compared and it was shown that significant differences in error probability result from using the two different signals with the same channel while very little difference in performance results from using the two different channels with the same signal.

For the three cases mentioned above, an approximate error probability expression developed by Sunde (3) was compared to the exact results. It was shown that this approximation does not provide a good estimate of the error probability when the frequency selective fading is of importance.

Next some experiments using computer simulations of communication systems subject to frequency selective fading were performed. The first of these was an investigation of the relative effects of the amplitude and phase distortion introduced in selective fading channels. It was shown that the phase distortion component has a deleterious effect on the system's performance while the amplitude distortion has a slight beneficial effect. These results provided the motivation for the development of the adaptive scheme mentioned previously for counteracting the effects of frequency selective fading. This system measures the residual delay and delay distortion introduced by the channel fading with pilot tones and then uses phase correction networks at the receiver to compensate for the measured phase characteristic. It was shown that this technique can result in error rate improvement factors of twenty or data rate increases by a factor of two to three. It was also shown that the system which corrects for both delay and linear delay distortion provides very little improvement over the system which corrects for delay only.

References

(1) Lindenlaub, J. C., and Bailey, C. C., "A Receiver System for Digital Communication over Frequency Selective Channels," Sixth Semi-Annual Research Summary, School of Electrical Engineering, Purdue University, June, 1967.

(2) Lindenlaub, J. C., and Bailey, C. C., "Digital Communication Systems Subject to Frequency Selective Fading," Report No. TR-EE67-17, School of Electrical Engineering, Purdue University, November, 1967.

(3) Sunde, E. D., "Digital Troposcatter Transmission and Modulation Theory," Bell System Technical Journal, Vol. 43, pp. 143-214, January, 1964 (Part I).

C. A Phase Lock Loop System with a Modulo $2n\pi$ Phase Detector

J. C. Lindenlaub

D. P. Olsen

The phase lock loop (PLL) shown in Fig. 1[1] has been modified slightly. The low pass filters in blocks 5 and 6 have been shifted to blocks 3 and 4 respectively to equalize the delay of the phase error components summed in block 12. The oscillator in Fig. 2[1] has been modified from a unijunction oscillator to a free running multivibrator to improve its phase jilts and frequency stability.

To facilitate the measurement of the loop acquisition time and the joint probability density function of the loop variables the test fixture block diagrammed in Fig. 1 was built. This PLL monitor makes it possible to measure the time duration of a large class of loop events. This system senses when a set of loop variables is in a prescribed subspace of the space spanned by these variables. This occurrence sets the upper flip flop thus closing the PLL and starting a time measurement. When another set or the same set of PLL variability is in a prescribed subspace the lower flip flop is set thus stopping the timer and opening the PLL. The control box determines whether or not this sequence will be automatically repeated or not. It provides for

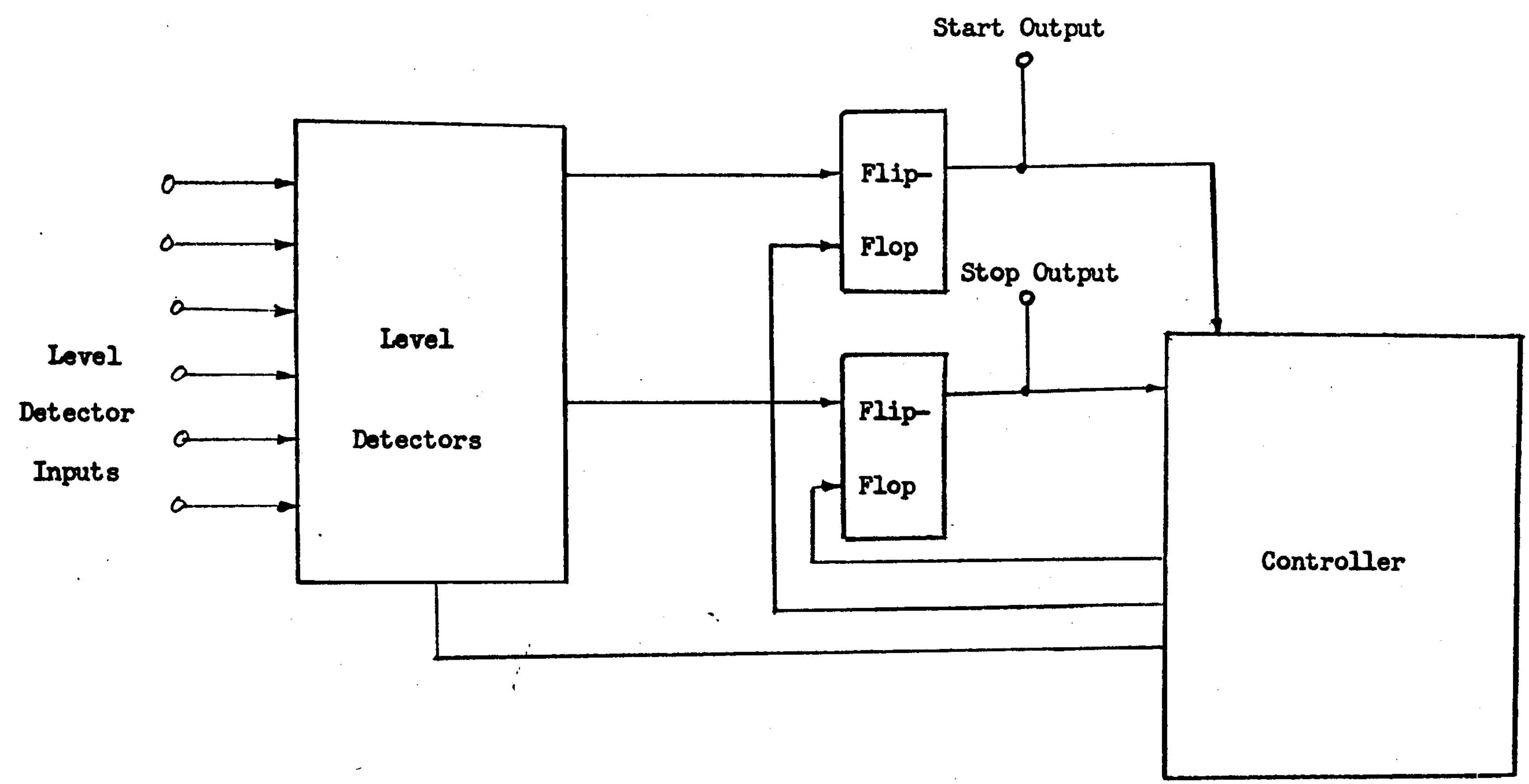

Fig. 1 Phase Lock Loop Monitor

manual specification of the subspaces for beginning and ending of the time interval. It provides the reset functions for the two flip flops or may be used to inhibit either flip flop's set input.

The theoretical analysis of the threshold characteristics of this phase lock loop has been continued. An attempt is being made to calculate the noise power in the loop output due to cycle slips of $2n\pi$ radians. It is felt that threshold occurs when this component of the output noise power becomes significant in comparison to the noise power predicted by linear analysis. This requires the calculation of the expected rate of $2n\pi$ radian cycle slips. Using Markov theory one can determine this from the expected rate of 2π cycle slips. Work is presently being done on this calculation.

References

(1) Sixth Semi-Annual Research Summary, Purdue University School of Electrical Engineering, June, 1967, pp. 81-83.

D. Source Encoding of Weather Satellite Data

P. A. Wintz

J. E. Essmann

During the past year we have investigated the use of complex exponentials as the basis functions for representing pictorial data. In particular, we have used these functions for cloud cover pictures obtained by tracking Essa II and Nimbus satellites. Some preliminary results are presented in Fig. 3. Fig. 3a shows an actual cloud cover picture after being time sampled at a rate of 800 Hz and reconstructed. Fig. 3b shows the same picture represented with only one fourth as many complex exponential samples. Note that the picture quality is better with the fewer complex exponential samples. It appears that we will be able to achieve data compression ratios of 10 - 1 by using this method. That is, for the same quality picture, only one tenth as many complex exponential samples are required than if time samples were used. The original picture (time reversed) is shown in Fig. 3c.

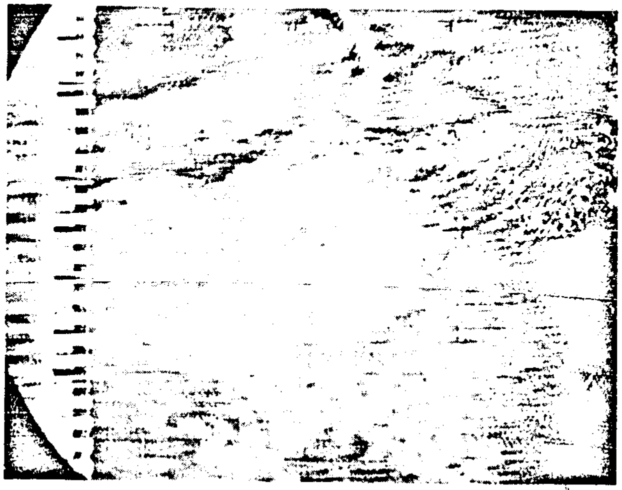

Figure 3a. 800 Time Samples Per Second

Figure 3b. 200 Exponential Samples Per Second

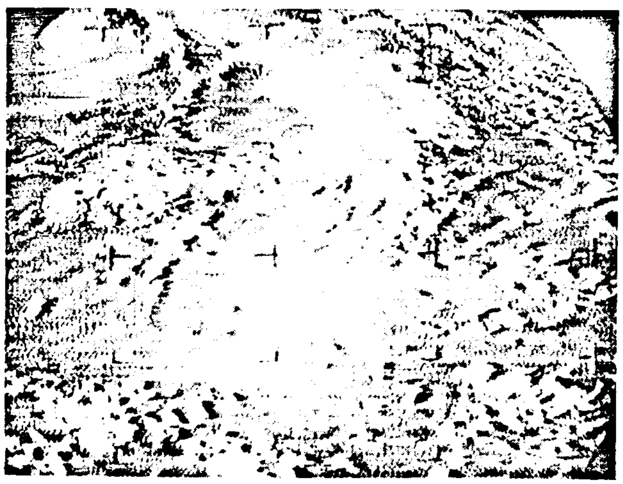

Figure 3c. Original Picture (Time Reversed)

FIGURE 3. CLOUD COVER PICTURES

E. Synchronization and Bit Timing

P. A. Wintz

E. J. Luecke

This project was concluded in December, 1967. A number of very significant results were obtained and will be reported in a Purdue University School of Electrical Engineering Technical Report TR-EE68-1 which will be available approximately February 15, 1968. The Abstract for this report follows:

In order to extract bit synchronizing pulses directly from equiprobable, binary, antipodal signals, nonlinear filtering of the bit train is necessary. This paper presents the results of experiments and analysis performed to determine the effects of the type of nonlinearity and of the shape of the symbol pulse on the quality of the extracted bit synchronizing signal and on the degradation of error rates of a correlation detector synchronized by these systems.

The performance of the optimum (maximum likelihood) and several commonly used suboptimum bit synchronizers are compared for pulse waveforms in additive Gaussian noise. Both the mean magnitude of the timing error (a measure of goodness of the bit synchronizer) and the probability of error (which takes into account the effect of timing jitter on the correlation detector) are used as performance measures. Both analytical and experimental results indicate a wide variation (10 db) in performances; the pulse shape has a significant effect.

The results indicate that considerably better quality synchronizing signals are obtained when the bit symbol is chosen to be a half sine or

raised cosine pulse instead of the commonly used square pulse. The results further show that when the half sine or raised cosine pulses are used, the performance with square law or absolute value nonlinearities is better than with an infinite clipper-differentiator combination.

Analysis of a correlation-type bit detector which is synchronized by these systems shows that probability of error for a fixed quality of synchronizing signal is degraded more with square pulses than with half sine or raised cosine pulses.

F. PCM Telemetry Systems - Theoretical Program

P. A. Wintz

A. J. Kurtenbach

This phase of the project was concluded in December, 1967. The significant results of this project are being published in the School of Electrical Engineering Technical Report TR-EE67-19 entitled "Analysis and Performance of PCM Telemetry Systems". The report will be available in mid-January. The Abstract follows:

Transmission of telemetry data is a significant application of pulse code modulation (PCM) communication systems. For analog data it is necessary for the system to sample, quantize and code the data before transmission over a digital channel. At the destination a facsimile of the source waveform is reconstructed by the receiver. It differs from the source waveform because of error introduced by the sampler, quantizer, and the noisy channel. To evaluate the effect of these errors on the reconstructed waveform we use the mean-integral-square-error criterion.

Our purpose is to minimize this error between sample functions of stationary, second order processes and their facsimiles presented to the user. Our approach is to tie together the existing specialized results adding necessary details and extensions where necessary in order to complete an analysis and optimization of the entire system.

We show analytically that the optimum quantizer is a function of the channel transition matrix and give the necessary conditions for a quantizer structure to be optimum. Suboptimum quantizer structures are also presented. Detailed numerical results are presented for the example of the stationary Gaussian Markov source with $R(\tau) = e^{-|\tau|}$ and the m^{th} order extensions of binary symmetric memoryless channels.

A data compression scheme using Hotelling's method of principal components on time samples is evaluated by computer simulation. It is found to perform comparably to Karhunen-Loeve sampling which is known to be best for this class of sources. For both of these sampling procedures the samples are found to have unequal variances and hence have varying information content. It is demonstrated that the samples with greater variance should be represented by more channel bits so as to minimize the overall system error. An ad hoc method of obtaining the unequal bit assignments is evaluated and shown to compare favorably to optimum bit assignments obtained by an exhaustive search.

A conventional PCM telemetry system which utilizes time samples, no data compression, and includes a noisy channel is simulated. In this manner an estimate of its overall performance is obtained. This is compared to the performance of the system using a K.L. sampler.

Both the system using a K.L. sampler and that using a time sampler followed by a digital filter are found to be superior to the conventional system. For the process considered the mean-integral-square error was decreased by 1/2 while obtaining a bandwidth reduction of 1/3.

When a digital filter is inserted after the time sampler it is also necessary to insert an inverse filter at the receiver. It is shown that the optimum filter is determined from the covariance matrix of the time samples.

G. PCM Telemetry Systems - Laboratory Program

P. A. Wintz

G. A. Apple, Jr.

During the past year we have designed and built a complete telemetry system. The sampler-quantizer, encoder, decoder, and digital-to-analog converter are constructed primarily from the plug in modules available from the Digital Equipment Corp. A signal generator and a noise generator are used for the modulators and channel, and a correlation detector was constructed using a Hewlett Packard mixer as an integrator. Dumping is accomplished with a diode bridge. During the coming year we plan to use this equipment to verify the theoretical results discussed in Section F.

H. Experimental Digital Filter

P. A. Wintz

A. Habibi

The approximation error in analog-to-digital conversion of a waveform for a given number of samples n, is minimized by expanding the waveform in terms of Karhunen-Loeve basis functions $\varphi_j(t)$; $j = 1,\ldots,n$. These functions are the eigenfunctions of the integral equation $\lambda\varphi(t) = \int_0^T R(t,\tau)\, \varphi(\tau)\, d\tau$ corresponding to n largest eigenvalues. The samples x_j are then the coefficients of the basis functions and are given by the equation $x_j = \int_0^T x(t)\, \varphi_j(t)\, dt$.

Instead of using a bank of analog filters with impulse responses of $\varphi_j(t-T)$ to compute to x_j, the above expression is approximated by the summation $x_j \doteq \sum_{i=1}^{m} x(t_i)\, \varphi_j(t_i)$. x_j's are determined by taking m time samples of the waveform x(t) in [0,T] interval and computing the summation. $\varphi_j(t)$ are computed from the integral equation by numerical techniques and are stored in a memory system. A fast digital multiplier has been designed and built for calculating the terms of the summation. The multiplier is built using integrated circuitry and has a multiplication time of less than 200 nanoseconds for six bit words. Increasing the number of bits to twelve would increase the multiplication time to about 400 nanoseconds.