Optimal Space Communications Techniques

Status Report

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Introduction

This status report summarizes several areas of research being supported under NASA Grant NGR-33-013-048 during the period December 16, 1969 to March 15, 1970. The topics covered in this letter are an All Digital Phase Locked Loop FM Demodulator, and an Optimum Adaptive Delta Modulator.

I. An All Digital Phase Locked Loop FM Demodulator

Up to this time, FM demodulation techniques have been unable to extend threshold more than several decibels. The second order phase locked loop was experimentally and theoretically shown to yield a 3dB threshold extension beyond the discriminator. It was theoretically shown, by Osborne and Schilling, that the third order phase locked loop could produce a 6dB or 7dB threshold extension. However, we have not been able to construct, using analog circuits, a third order system to achieve the predicted performance levels. It has been conjectured that the reason for the failure in implementation is the presence of extraneous poles in the system function, and the nonlinear characteristics of our devices.

To avoid these problems, we have begun the design of a phase locked loop using digital circuitry: digital multipliers, adders, and filters. A digital voltage controlled oscillator has been designed. In using a digital synthesis procedure we are eliminating the unknown poles and unknown nonlinear characteristics from our system. However, we are replacing these problems by others, which are common to digital systems. These include: quantization noise, roundoff error and overflow error.

One very interesting problem, which we have just solved, arises from the multiplier present at the phase locked loop input. In an analog system we multiply the incoming signal and noise by the output of the voltage controlled oscillator. The product consists of a difference-frequency and a sum-frequency component. A low-pass filter eliminates the sum-frequency component. In a digital system, we first convert the incoming signal and noise into a PCM waveform. The output of the voltage controlled oscillator, also a PCM waveform, is digitally multiplied. by the PCM equivalent of the input. The resultant waveform

when digitally filtered is periodic with respect to the sampling time. Unless extreme care is taken the sum-frequency term is not attenuated and is shifted to the same frequency band as the difference-frequency term. If this occurs the phase locked loop cannot be constructed. We have determined the correct sampling rate to insure that the digital filter will attenuate the sum-frequency term.

We are also currently investigating the use of other multiplier configurations such as one having a sawtooth characteristic, so that we can determine which configuration yields the greatest threshold extension.

Future Work

1. To complete the design of a digital phase locked loop, and to construct it.
2. To determine theoretically and verify experimentally the response of the phase locked loop to an FM signal.
3. To design, construct, and test higher order loops.
4. To investigate the use and limitations with time division multiplexing.

II. An Optimum Adaptive Delta Modulator

In a digital communication system we very often encounter the problem of converting a source sequence into binary digits. Delta modulation is a simple source encoding-decoding technique. The advantage of delta modulation over pulse code modulation is its extremely simple circuitry which is easily integrated on a single ship. This provides a system having a small volume and low weight, two desireable characteristics for NASA's systems.

A major disadvantage of a delta modulation system is its limited dynamic range. At low input signal power levels the step-size produced by the delta modulator is too large. The error produced oscillates between positive and negative values as though no signal were present. Thus the SNR decreases rapidly as the input signal power decreases. If, on the other hand, the input signal power is too large, then during a time interval where the input signal looks like a "ramp", the error signal will always be positive. The large error results in an increased noise and again, the SNR

decreases rapidly, this time as the signal power increases. The first effect is due to "quantization error" and the second due to "slope overload".

To achieve a wider dynamic range we must adapt the step size to some estimate of the signal power and slope. One of the more popular techniques of step size adapting is Abate's procedure. In this procedure, the step size is increased in a prescribed manner as the number of adjacent positive (or negative) errors increases. The step size is reduced if adjacent errors are of opposite polarity. Abate's procedure results in a dynamic range of approximately 30-40dB as contrasted to 5-10dB dynamic range for the ordinary (non-adaptive) delta modulator.

We have obtained an explicit expression for the optimum step size as a function of the polarity of N past errors. This general result was then specialized to the case of the polarity of 2 past errors. This specialization resulted in less than 0.5dB degradation in SNR. The dynamic range of the optimum adaptive delta modulator using 2 past terms greatly exceeds 80dB.

The feedback mechanism for the ordinary delta modulator is a simple integrator, or RC low pass filter. The feedback system required for the optimum adaptive delta modulator is extremely complicated, requiring ratios of Q-functions having arguments which are functions of the past samples of the error polarity. Fortunately, we have been able to approximate this highly nonlinear feedback system by a piecewise-linear system of the form

$$f(\bullet) = \begin{cases} 0 & (\bullet) < -1 \\ 2 + 2(\bullet) & -1 < (\bullet) < 5.4 \\ 0 & (\bullet) > 5.4 \end{cases}$$

As a result, we can construct our delta modulation system. We are currently designing the encoder and decoder systems.

The linearized feedback system has been simulated on a digital computer. We have found no decrease in the dynamic range of the system. However, we have found a decrease in SNR results. This decrease is less than 1dB for all practical sampling rates.

Future Work

1. To find the response of the optimum adaptive delta modulator to thermal noise, which results in bit errors in the receiver. This analysis is possible, without relying on computer simulation, because of the piecewise-linear approximation of the feedback loop.

2. To construct the optimum adaptive delta modulator and determine experimentally the output SNR due to quantization and thermal noises. These results will then be compared to the theoretical results.

Papers Published

Two papers on the various aspects of this problem have been approved for presentation:

1. "Determination of the Optimum Step Size of a Delta Modulation System", C. L. Song and D. L. Schilling, to be presented at the 1970 International Information Theory Symposium in the Netherlands.

2. "An Optimum Adaptive Delta Modulator", C. L. Song and D. L. Schilling, to be presented at the 1970 International Conference on System and Circuit Theory in Kyoto, Japan.

Mr. Song is Dr. Schilling's doctoral student.