Spread Spectrum Radio Networks: Final Report

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Final Report

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packet radio networks, frequency-hop packet radio, error-control coding, time-hopping.

This is the final report for Contract DAAG29-84-K-0088 (ARO Proposal 21576-EL) for the period June 22, 1984 to July 20, 1987. A list of publications is attached.
This report reviews the accomplishments in the research program entitled Spread Spectrum Radio Networks, supported by the U.S. Army Research Office under contract number DAAG29-84-K-0088 (proposal number 21576-EL), which was for the period June 22, 1984 through July 20, 1987. This research was carried out at the University of Illinois under the direction of Professors M. B. Pursley and D. V. Sarwate who were the principal investigators for the contract.

**Soft-decision decoding of RS codes**

One phase of our research program was concerned with soft-decision decoding for Reed-Solomon codes with application to both anti-jam and multiple-access frequency-hop spread spectrum. Modifications of Forney's generalized minimum distance (GMD) decoding technique were developed to give an algorithm which performs better than the original algorithm. The performance analysis carried out by Forney for his algorithm is for coherent binary antipodal signaling in white Gaussian noise. His results showed that for large signal-to-noise ratio, the asymptotic performance of his algorithm is equivalent to maximum likelihood decoding. The performance at low signal-to-noise ratios, however, was found to approach that of errors-only decoding. Because applications to anti-jam and low-probability-of-intercept communication require operation at low signal-to-noise ratios, it is of considerable importance to obtain better performance in this region. We have obtained performance estimates for the modified algorithm that show improved performance over a wide range of signal-to-noise ratios.

We have analyzed an improved version of GMD decoding for binary orthogonal signaling on an additive white Gaussian noise channel. This has been used to compare the improved version with GMD decoding. Initially, we found an upper bound for the codeword error probability of the improved version of GMD decoding, and compared it to Forney's upper bound for GMD decoding. The results looked promising, but were inconclusive since we were comparing upper bounds.

We have now utilized characteristic functions in a way that permits us to find the exact codeword error probabilities for both the improved algorithm and GMD decoding.
for the channel cited above. The results show that for low to moderate signal-to-noise ratios, GMD decoding performs insignificantly better than errors-only decoding. In contrast to this, we found that the performance of the improved algorithm is about 1 dB better over the same range of signal-to-noise ratios. A trivial extension of Forney's work shows that for the channel in question, maximum likelihood decoding will perform only about 3 dB better as the signal-to-noise ratio increases. This means we should expect that 3 dB is the best we can do for minimum distance decoding schemes, hence a 1 dB improvement is significant.

The GMD decoding technique is carried out by a sequence of attempts at errors and erasures decoding. Each successive decoding is performed on received data for which the two least-reliable, unerased symbols are declared to be erasures. For Reed-Solomon codes, this requires multiple applications of some method for finding the error locator polynomial. We have developed an improved search technique in which information from the present decoding attempt can be used to reduce the computation needed in the next step of the iteration.

Performance of frequency-hop radios in a multiple-access environment

We have obtained several important results on the use of frequency-hop transmission for radio networks. We have demonstrated the performance gains that can be obtained by employing frequency-hop transmission, error-control coding, and side information in such a network. The assumption in our initial work is that side information, when available, is perfect. That is, the decoder knows which of the received symbols have been hit by multiple-access interference. In addition, the initial research ignored the effects of background noise, such as thermal noise in the receiver.

In the actual system, the side information is not perfect and there will be background noise. Thus, the decoder will occasionally be informed that a particular symbols has been hit when, in fact, it has not, and it will sometimes be informed that a symbol has not been hit when, in fact, it has. The former events are referred to as false alarms and the latter are misses. There are two issues that were resolved in order to
demonstrate the practical applicability of the use of side information in spread-spectrum radio networks.

First, we determined how large the false alarm and miss probabilities can be before the performance of the system is unacceptable. Second, we developed techniques to derive this side information in the receiver. We also completed an investigation of the tradeoffs between false alarms and misses, and we analyzed the performance of several schemes for obtaining side information in practical spread-spectrum radios.

The performance results for a side-information system that employs the transmission and detection of test symbol patterns demonstrate very clearly the benefits of side information in frequency-hop transmission. In fact, this method produced lower false alarm and miss probabilities than we had anticipated. Based on our preliminary results, it seems that no other mechanism (e.g., AGC outputs) for generating side information will be needed for a slow-frequency-hop spread-spectrum system. Adequate reliability can be obtained from the inclusion of test symbols or parity check symbols in the message or packet. Recent work has been devoted to the evaluation of packet error probabilities in a multiple-access environment when this, or any other, method for generation of side information is employed. The analytical difficulties are due to the strong dependence between the interference for symbols in the same dwell interval. This has been dealt with successfully, and several conference papers based on this work have been published. Several journal articles are being prepared for submission in the very near future.

Spread-spectrum packet radio with convolutional coding and Viterbi decoding

We developed a new bound on the packet error probability for spread-spectrum radios which use binary convolutional coding, hard-decision demodulation, and Viterbi decoding. This bound is a general result for convolutional codes with Viterbi decoding; it is not limited to a particular modulation technique. Applications to both direct-sequence and frequency-hop spread-spectrum radio were considered, and the results of our investigation allowed us to compare the performance of these two forms of spread spectrum when employed in a multiple-access communication network environment.
Our work on adaptive encoding and decoding for convolutional codes for spread-spectrum communication networks is still in progress. This research focuses on adapting the code rate and other aspects of the encoding and decoding to the channel traffic. We are considering both rate 1/n and rate (n-1)/n convolutional codes, as well as various rate Reed-Solomon codes. High-rate codes are included for use in light traffic and interference, and low-rate codes are for heavy traffic and interference. The primary difficulty is due to the lack of analytical methods for the evaluation of the performance of Viterbi decoding for large values of the channel error probability. For large error probabilities on the channel, both the union bound and the usual truncation procedure for the state-diagram transfer function cause large deviations from the true error probability. We have methods for overcoming the problems with truncating the transfer function, but there are no good analytical methods for overcoming the problem with the union bound. Additional work is in progress on this topic.

**Delay in frequency-hop packet radio networks**

This project is concerned the delay characteristics of frequency-hop packet radio networks as a function of the channel access and retransmission protocols. The basic problem is that the number of successful packet transmissions for a given packet slot is a random variable, and it is very difficult to determine the distribution function for this random variable. The difficulty arises from the fact that packet successes are strongly dependent, so this distribution function can be calculated exactly for very small networks only.

Our approach is to combine the use analytical results with results obtained from a computer simulation in order to verify the approximation and gather information for situations in which the analysis is not applicable. The simulation tracks the progress of a single packet in a fully connected multiple-access network until the packet is either successfully received or it is discarded. Although this simulation does not include the effects of thermal noise, a second simulation has been written which tracks the progress of all packets that enter the network and this simulation does include the effects of both
thermal noise and multiple-access interference. Incremental redundancy is compared with the traditional retransmission approach by the second simulation.

In the traditional approach, a retransmission consists of a duplicate version of the original transmission, but for incremental redundancy transmission, the second and successive transmissions consist of additional redundant symbols that are used in conjunction with all previous transmissions to decode the message. By use of incremental redundancy, the error and erasure correcting capability of the code is increased with each retransmission, and the packet is correctly decoded when enough redundancy is available to correct the errors and erasures. We have obtained very dramatic reductions in the delay along with an increase in throughput by using incremental redundancy.

The simulation study has been completed, and a paper based on the results has been submitted for publication. The channel access protocol employed in the network is slotted ALOHA, so that all packet receptions are entirely contained within the time slots. The simulation includes the effects of fixed delays to account for propagation between terminals and processing at the receiving terminals, and a feature is included to discard packets after a certain number of unsuccessful attempts to transmit the packet. The model is one in which there are a finite number of channels available to the network for packet transmissions, and three different methods are considered for the use of these channels. One method includes frequency-hop transmission as a special case. It is assumed in all three methods that a given terminal may transmit on at most one channel at a time, but the three methods differ in the degree to which they permit a packet transmission to use multiple channels. Typically, the total number of terminals in a network is much greater than the number of channels, but only a fraction of the terminals are transmitting in a given time slot.

In the simplest scheme, each terminal is assigned a specific channel, and all of its transmissions must take place on this one channel; this scheme is referred to as fixed-assignment, fixed-channel transmission. In a second scheme, each terminal makes an independent random selection of a channel for each time slot. If a terminal has a
packet to transmit in a particular time slot, it sends the entire packet over the channel that corresponds to that time slot, so this scheme is referred to as random-assignment, fixed-channel transmission. In the third scheme, the time slots are divided into L subintervals, and each terminal makes an independent random selection of a channel for each subinterval. Each packet is divided into L subpackets, with a fixed number of channel symbols in each subpacket, and each of these subpackets is transmitted in the corresponding subinterval. Thus, in this scheme, which we refer to as channel-hopping transmission, a given packet may be transmitted over as many as L different channels, depending on the outcome of the L independent channel selections for that packet.

The primary example for this setting involves the division of the network’s allotment of the RF spectrum into L equal-bandwidth frequency slots. In the context of this example, the first scheme corresponds to classical frequency-division multiple access (FDMA), while the second corresponds to making a random selection of the frequency slot to be used for a given packet, and then transmitting the entire packet over that frequency slot. The third scheme corresponds slow-frequency-hop transmission which is described in detail in our previous publications.

Our results show that for the cases considered, channel-hopping gives lower maximum throughput than the other two schemes if the propagation and processing delays are small, but it gives higher throughput and smaller average delay if the propagation and processing delays are large. Studies of the code rate were also carried out, and these show that, for most situations, a code rate of 3/8 (e.g., a (32, 12) RS code) is superior to rates of 1/2 or 1/4, which is consistent with our previous analytical findings.

Work is also in progress on a simulation of FH packet radio networks which are not fully connected (i.e., relay networks, also known as multi-hop networks). The goal is to study the effects of several different protocols on the network throughput and delay. Particular emphasis is being placed on the pacing and alternate routing protocols in our early investigations with this simulation.

New analytical results have also been obtained for fully connected FH packet radio networks. This area of theoretical research is concerned with asymptotic limits on
throughput for heavy traffic, large bandwidth, and long block length Reed-Solomon codes. The primary results are loosely described as follows. The performance of FH systems is relatively sensitive to the distribution of packet lengths. Use of the slotted ALOHA transmission protocol does not improve the asymptotic performance in an asynchronous FH packet radio network. The capacity of the FH channel can be attained using Reed-Solomon coding under Poisson traffic conditions.

**Time- and frequency-hopping for random access communication**

We have completed several studies of time-hopping and frequency-hopping systems for random access communications. A new scheme had been analyzed in which each terminal is capable of transmitting and receiving in n frequency bands simultaneously. We have shown that with a judicious assignment of frequency bands to transmitters and receivers, and with the use of an interleaved (n,k) Reed-Solomon code, it is possible to achieve k times as much normalized throughput as the slotted ALOHA scheme. This increased throughput is possible at low traffic rates only, while at higher traffic rates, the slotted ALOHA scheme has larger throughput. However, low traffic rates must be maintained if one desires to achieve the low packet erasure rates generally required in military communications. Different versions of this scheme have different properties. When bandwidth expansion is feasible, the design of the receivers and transmitters is simplified, while the packet delay is the same as that of a comparable narrowband slotted ALOHA scheme. On the other hand, if the bandwidth cannot be increased, then additional delay must be tolerated. However, the synchronization requirements are no longer as stringent as in the case of slotted ALOHA.

We have continued our study of the construction of time-hopping patterns for random access communications. The results obtained include a method of constructing n-hop patterns (that is, those which use n time slots) which span fewer than n^2 slots for all n such that either n+1, n, n-1, or n-2 is a prime power. Since 35 is the smallest value of n for which none of n+1, n, n-1, or n-2 is a prime power, this shows that such patterns exist for all n ≤ 34. By actual construction, we have shown that such patterns exist for all n ≤
This provides time-hopping patterns for all practical purposes.

Previously, we showed that for any fixed number of time-hops and given level of offered traffic, a code rate of approximately $1/3$ maximizes the throughput provided the offered traffic is not too large. This had also been observed from numerical data. For any fixed level of offered traffic and fixed number of data transmissions, increasing the number of redundant (or parity check) transmissions beyond certain limits actually reduces the throughput. This is what one might intuitively expect on the grounds that the additional transmissions would clutter up the channel without necessarily increasing the probability of correct packet transmission significantly. We have since analyzed a number of frequency-hop schemes and hybrid time-hopping and frequency-hopping schemes using a variety of different hopping techniques, and find that the code rate which maximizes the throughput is approximately $1/3$ for all of them.

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Theses


A. Krishna, “Negacyclic MDS codes of length \( q+1 \) over GF(\( q \)),” M. S. thesis, Department of Electrical and Computer Engineering, July 1987; D. V. Sarwate, advisor.


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