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Abstract

Results are reported for basic research in mobile wireless communication networks for tactical applications including investigations of new methods for error-control coding and decoding, modulation and demodulation, channel access, adaptive transmission and routing, and broadband antenna design. Research results are presented on adaptive, energy-efficient, distributed protocols for mobile wireless networks that must operate effectively over unreliable communication links in highly dynamic environments and handle multimedia traffic. The dominant feature of the research is the exploitation of interactions among protocols to capitalize on the opportunities and overcome the impediments presented by the tactical communications environment and capitalize on the differences in quality-of-service requirements for mixed-media traffic. The interactions among protocols involve not only the exchange of information but also the active cooperation of different classes of protocols to accomplish the common objective of reliable, energy-efficient distribution of information. The research accomplishments include establishing and taking maximum advantage of a strong coupling of the various protocol layers with physical-layer functions such as receiver processing, modulation and demodulation, error-control coding and decoding. New protocols have been developed and evaluated for directional antennas, and broadband antennas have been designed and tested for use with spread-spectrum communications and other wideband waveforms. The development of side information in physical-layer operations and its effective utilization in soft-decision decoding, adaptive transmission, and adaptive routing are fundamental elements of the communication techniques and adaptive protocols that have been designed and evaluated.
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14. ABSTRACT  
Results are reported for basic research in mobile wireless communication networks for tactical applications including investigations of new methods for error-control coding and decoding, modulation and demodulation, channel access, adaptive transmission and routing, and broadband antenna design. Contributions include adaptive, energy-efficient, distributed protocols for mobile wireless networks that must operate effectively over unreliable communication links in highly dynamic environments and handle multimedia traffic. The dominant feature of the research is the exploitation of interactions among protocols to capitalize on the opportunities and overcome the impediments presented by the tactical communications environment and capitalize on the differences in quality-of-service requirements for mixed-media traffic. The interactions among protocols involve not only the exchange of information but also the active cooperation of different classes of protocols to accomplish common objectives. New protocols have been developed and evaluated for directional antennas, and broadband antennas have been designed and tested for use with spread-spectrum communications and other wideband waveforms.

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Research Results

In research on slow-frequency-hop (FH) transmission, we examined three classes of modulation: binary orthogonal modulation and noncoherent demodulation, binary antipodal modulation with coherent demodulation, and nonbinary quadrature amplitude modulation (QAM) with coherent demodulation. When compared with standard binary or quaternary modulation, QAM provides a greater spectral efficiency at a cost of a larger required signal-to-noise ratio to meet a specified error rate. For FH spread-spectrum, judicious use of error-control coding can decrease the signal-to-noise ratio requirement and also provide protection against partial-band interference. We considered two classes of turbo codes for use in FH spread spectrum with QAM FH: serially concatenated convolutional codes with iterative decoding and block product codes with iterative decoding (i.e., turbo product codes). In investigations of QAM conducted jointly with MIT Lincoln Laboratory [PPP04], we found that for both the log likelihood ratio metric and our new low-complexity distance metric, the turbo product codes and serially concatenated convolutional code give much better performance than the binary convolutional code that is the industry standard for IEEE 802.11. Each of the turbo codes gives lower packet error probabilities and higher throughput than the convolutional code for FH transmission over an additive white Gaussian noise channel. In addition, the turbo product code is superior to the convolutional code for channels with partial-band jamming. Performance comparisons are given in [PPP04] for 16-QAM with the two types of turbo codes and soft-decision metrics based on the log likelihood ratio. The performance results for the turbo codes are compared with the performance of the IEEE 802.11 standard convolutional coding and Viterbi decoding for 16-QAM. An alternative soft-decision metric for Gray-coded M-QAM is developed and evaluated for each of the three coding systems.

High-rate direct-sequence spread spectrum is a modulation technique in which most or all of the spreading is provided by nonbinary coded data modulation. For applications to mobile ad hoc wireless networks, the limited processing gain of high-rate direct sequence spectrum gives only modest protection against multipath and multiple-access interference, so its use is restricted to fairly benign channels. In [PuR04], we explore the increased interference rejection capability that can be obtained from the inclusion of error-control coding. The three forms of error-control coding that we have investigated are convolutional coding with Viterbi decoding, Reed-Solomon coding with errors-and-erasures decoding, and turbo product coding with iterative decoding. We showed that error-control coding permits the use of high-rate direct-sequence spread spectrum, even on channels with fairly strong multiple-access interference. Soft-decision decoding
metrics that distinguish among the bits in a symbol (i.e., bit metrics) give the better performance than those that assign the same weight to each bit in a symbol (i.e., symbol metrics). Each of the three forms of coding provides significantly greater multiple-access capability than is obtained if the system has no error-control coding. Commercial implementations of the encoders and decoders are available, so the performance gains that we demonstrated can be achieved with little additional system development time. Of the three types of codes that we examined, turbo product codes give the best performance. The choice between convolutional codes with Viterbi decoding and Reed-Solomon codes with errors-and-erasures decoding depends on the application and the metric. Adaptive transmission protocols can give large performance improvements for high-rate DS spread-spectrum communication over time-varying channels, and the key parameters for the adaptive protocols can be determined analytically by using the methods in [PuR05a].

Alternative protocols are presented in [PuS05] for adapting the code from packet to packet in a mobile, wireless, ad hoc FH network with FH transmission, coherent or noncoherent demodulation, and soft-decision decoding on channels with time-varying partial-band interference. The protocols are implemented in subsystems that are external to the demodulator and soft-decision decoder, so they can be employed with a wide range of codes and decoding algorithms. No channel measurements are needed and no external side information is supplied to the receiver. Protocol performance is compared with capacity limits and with an ideal protocol that uses perfect channel state information for the most recent packet transmission and selects the code that maximizes the expected throughput. Our decoding subsystem adaptively scales the soft decisions at the input to the decoder. The information needed to adapt the scale factor for a given dwell interval is derived from the demodulation of the received code symbols in the dwell interval. In order to minimize the response time, the choice of the code for a packet depends only on the scale factors for the previous packet reception. The interference is modeled as partial-band noise that occupies an unknown fraction of the band. The partial-band noise model is convenient for use in analysis and simulation, and it represents a good approximation for many frequency-hop systems with partial-band interference. For a dynamic channel, the unknown fraction of the band occupied by partial-band interference is constant for the duration of a packet, but it may change from one packet to the next. In the model used for our performance evaluations, changes in the fraction are governed by a finite-state Markov chain whose states and transition probabilities are unknown to the transmitter and receiver.
The protocol for adaptive coding that is described and evaluated in [PuS05] is suitable for most binary codes and soft-decision decoders. In particular, the protocol can be used with binary convolutional coding or any form of binary turbo coding. Although our numerical results are for turbo product codes, most of our adaptive methods are applicable to other codes and other decoding algorithms. Our results demonstrate the feasibility of the adaptive coding technique and the performance advantages of adaptive coding for channels with time-varying interference. In particular, the adaptive coding system obtains the maximum possible throughput when there is little interference and it provides higher throughput than fixed-rate coding when the interference occupies a larger fraction of the band. The performance of our protocol is also close to the upper bound that corresponds to an ideal protocol with capacity-achieving codes.

Direct-sequence (DS) and FH spread-spectrum modulation can combat fading that limits performance in mobile packet radio communications. When used in conjunction with coding and interleaving, FH spread spectrum provides diversity by taking advantage of the frequency selectivity of the fading. DS spread spectrum provides diversity by permitting individual multipath components in the received signal to be resolved and combined in the receiver. These two forms of diversity result in different performance characteristics for the two modulation formats. The difference is examined in [GNP02], where FH and DS systems of comparable complexity are defined and their performance is compared over several doubly selective fading channels. The probability of packet error is determined for each system for a single transmitter-receiver pair. Several propagation environments are considered that are modeled by channels with different multipath delay profiles. The effects of the carrier frequency and the relative velocity between the transmitter and the receiver are examined by considering a range of Doppler spreads.

It is shown in [GNP02] that the performance of the FH system is insensitive to variations in the Doppler spread of the channel. The time selectivity of the channel does not significantly increase the level of diversity that can be obtained using FH spread-spectrum modulation. In contrast, the level of diversity that is obtained with DS spread-spectrum modulation depends heavily on the channel's time selectivity. Thus the performance of the DS system is very sensitive to the Doppler spread of the channel, and much better performance is obtained for large Doppler spreads. Consequently, for a given channel and packet size the performance of the FH system is best relative to the performance of the DS system if the Doppler spread of the channel is small, and the relative performance of the DS system is best if the Doppler spread is large. Our results also show the relative performance of the DS system is best if most of the power in the
received signal is concentrated in a small number of resolvable specular or diffuse multipath components. If the signal power is spread evenly across a wide range of path delays, the FH system exhibits performance superior to that of the DS system.

Part of our research on the use of radios with adaptive arrays in distributed packet radio networks has focused on the performance and design of the algorithm for acquisition of a DS (DS) spread-spectrum packet transmission. The DS packet radio system is assumed to employ a common acquisition preamble for each transmission. The preamble is obtained from a long-period pseudorandom generator with a time-based algorithm that determines the preload for the generator (and thus the preamble sequence). If the seed for the time-based preload-generation algorithm is known only to the radios in the network, a measure of security and low probability of interception are achieved. We have considered receivers with omnidirectional or switched, sectored antennas. We have evaluated the performance of a noncoherent, serial acquisition technique for DS spread-spectrum packet communications [NRB01]. The acquisition technique that is considered uses threshold crossing of a matched-filter output to detect the fixed-length preamble at the start of each packet. We first consider the use of a fixed threshold that does not change as the preamble sequence changes. The analysis accounts for differences in the frequency references of the transmitter and the receiver as well as Doppler shifts resulting from the mobility of the radios. It also accounts for the effect of the intermediate-frequency filter and the subsequent AGC system. The role of the AGC system in determining the acquisition performance is examined. The complex interdependence among the AGC system, the channel conditions, and the acquisition threshold in determining the performance of the acquisition technique is illustrated by several examples for which the probability of not acquiring is a markedly non-monotonic function of the signal-to-noise ratio.

We have extended the investigation to consider a serial acquisition algorithm in which the receiver samples the output of the matched filter at twice the chip rate during acquisition and in which the signal is tracked with a delay-locked loop (DLL) after acquisition [Non01b]. It is shown that the sample-timing error in the receiver has a significant effect on both the acquisition performance and the severity of the non-monotonicity that it exhibits as a function of the signal-to-noise ratio. Consequently, accounting for random sample-timing error can substantially affect the choice of the acquisition threshold. The effect of pull-in range of the DLL on acquisition performance is also considered, and it is shown that the performance is not sensitive to the pull-in range as long as it is at least twice the chip duration.
We have also extended the investigation to consider a serial acquisition algorithm in which the acquisition threshold is adapted based on the autocorrelation properties of the preamble sequence that is currently in use [SwN02]. It is shown that for a given preamble sequence and a well-chosen fixed threshold, the worst-case acquisition performance over a range of values of SNR may occur at the lowest SNR, the highest SNR, or an intermediate SNR, depending on the sequence. And it is shown that some sequences have sufficiently poor autocorrelation properties that acceptable performance cannot be obtained over any reasonable range of SNR with any given threshold. If instead the adaptive-threshold technique is used, a much smaller subset of preamble sequences results in unacceptable acquisition performance, and the average acquisition performance over all sequences is improved substantially.

We have also developed and evaluated the performance of a serial acquisition algorithm which employs an estimator to adaptively select the acquisition threshold on a sample-by-sample basis [SwN03, SwN05]. The estimator is a windowed linear filter applied to the recent past samples out of the matched filter of the acquisition stage. It is shown that the filter output is an unbiased estimate of the mean sample value if only noise is present, and it has a small negative bias is a signal is present. It is also shown that the technique reduces the severity of the non-monotonicity, and indeed it achieves a nearly constant false-alarm rate over an arbitrary range of (unknown a priori) signal-to-noise ratios. The technique substantially improves the acquisition performance in comparison with both the fixed-threshold technique and the first adaptive-threshold technique. It is also shown that the second adaptive-threshold technique results in a substantial reduction in the sensitivity of the performance to variations from the nominal bandwidth of the IF noise-rejection filter, and it eliminates sensitivity to variations of the AGC's steady-state output voltage from its nominal value. We have examined the second adaptive-threshold technique for both serial and hybrid serial/parallel algorithm for acquisition of DS packet transmissions [Non01a, NoS02]. It is shown that the techniques works well with both algorithms, and the hybrid algorithm results in better performance than noncoherent serial acquisition.

We have developed and evaluated the performance of a multichannel RTS-CTS-based channel-access protocol in a DS spread-spectrum packet radio network that contains an arbitrary mix of nodes with multiple directional antennas and nodes with omnidirectional antennas [SNR04]. We demonstrate a counterintuitive phenomenon that occurs due to differences in channel-state information between the transmitting node and the receiving node. The phenomenon can result in poorer network performance in some circumstances if some of the nodes use multiple directional antennas than if all the nodes use...
omnidirectional antennas. We have shown that the phenomenon is a generalization of the receiver blocking problem that has been identified previously in ad hoc networks in which all the nodes have omnidirectional antennas. However, as we demonstrate, this phenomenon can have much greater impact on performance if the transmitting and receiving nodes have directional antennas. We examine in detail the mismatch in channel-state information that is the underlying reason for the receiver blocking problem. And we demonstrate a modification of the multichannel RTS-CTS protocol that corrects this mismatch in channel-state information by exploiting the availability of the control channel. We show that this modified protocol substantially mitigates the receiver blocking problem, and it results in much better performance in a networks in which some of the nodes employ directional antennas.

We have also developed and evaluated three variations of an RTS/CTS channel-access protocol that is designed to improve the spatial efficiency of traffic-channel utilization (and thus the network performance) in a heterogeneous DS packet radio network [SNR05]. In addition to use of an RTS/CTS exchange to negotiate a link transmission, the protocol includes a mechanism by which the negotiating pair determines which in-use traffic channels should be treated as reserved in the local area and which ones are available for concurrent use. The performance of each variant of the new protocol is compared with the performance of two existing RTS/CTS protocols which employ a static policy concerning traffic channel reuse. (One of the existing protocols employs exclusive reservation of a traffic channel within the local area, and the other existing protocol places no restrictions on concurrent use of each channel.) It is shown that the dynamic approach of the new protocol results in much better performance than either protocol using a static approach. Several network topologies are considered. With a low chip rate, the new protocols yield performance improvements of 20-70% over the exclusive-use protocol and 70-130% over the unrestricted reuse protocol. With a high chip rate, the new protocols give performance improvements of 40-55% over the exclusive-use protocol and provide comparable performance to the unrestricted-use protocol. Moreover, the new protocols provide performance that is much more robust over varying chip rates.

The TCP protocol does not perform well in a connection that includes one or more lossy links. Yet it is used in much of wide-area DOD communications, and thus a connection that spans a wide-area network and a wireless subnetwork is likely require support of TCP connection with one endpoint in at the wired network terminal. These requirements are effectively balanced by an approach referred to as "split-connection TCP" in which the TCP connection spans only the wired subnetwork and the wireless subnetwork using
another transport protocol that is more amenable to the characteristics of wireless communications. Previous research on split-connection protocols has used simulations exclusively. In [XHN04], we develop an analytical model of a split connection that is sufficiently general to incorporate a wide range of circumstances within the wireless subnetwork. The model allows for closed-form evaluation of the connection's steady-state throughput under conditions of heavy traffic, and it accounts for wireline propagation delay and the size of the buffer allocated to the connection in the wireline-wireless gateway node. The model's use is illustrated in [XHN04] by considering the problem of sizing the buffer in the gateway node. It is used in [XHN03] to consider the performance of a single wireless hop at one end of the connection. The impact of the wireless-link ARQ protocol on connection throughput is evaluated. It is also shown that the delay in the wireline network does not affect the heavy-traffic throughput if the connection's buffer in the gateway node is sized properly.

We designed, implemented in simulation, and evaluated an energy-efficient routing protocol for frequency-hop wireless networks, and the results are summarized in [PRW02a] and [PRW04b]. The link resistance measures provide a protocol that accounts for energy consumption and link quality in the selection of routes in a wireless network that exhibits time varying propagation losses and interference. Several resistance measures have been devised, and performance comparisons have been completed. Tradeoffs among energy efficiency, delay, and packet success probability have been investigated. Our results demonstrate that if the routing protocol focuses primarily on energy conservation, the delay and end-to-end success probability may be unacceptable for many types of traffic. Our results also demonstrate that a compromise between energy conservation and delay performance can be achieved by the use of a hybrid quality-energy resistance measure. This resistance measure provides a tradeoff among throughput, energy conservation, delay, and end-to-end success probability.

The use of adaptive-transmission protocols in wireless, store-and-forward, packet communication networks may result in large differences in the energy requirements of the alternative paths that are available to the routing protocol. Carefully designed routing metrics use physical-layer information to assign quantitative measures of the quality and energy efficiency of the paths from the source to the destination. Such measures are required if the routing protocol is to take advantage of the potential energy savings that are made possible by the adaptive-transmission protocol. The first integrated energy-efficient protocol suite for routing and adaptive transmission in FH wireless networks is described and evaluated in [PRW04b]. Several routing metrics are compared, and tradeoffs among energy efficiency, delay, and packet success probability are investigated.
We show that routing protocols designed only to minimize energy give poor performance, and we established that cross-layer protocols are essential for energy-efficient routing with acceptable QoS.

Conservation of energy is important in a packet radio network, but reliance on energy-efficient routes for all types of traffic leads to large delays and low packet delivery probabilities in many situations. For mixed-media traffic, QoS priorities vary among the traffic types, and the protocols must account for the differences in these priorities (e.g., low delay for voice traffic, error-free reception for computer files). Our energy-efficient multimedia protocols described in [PRW04a] and [PRW05] account for these differences and select routes according to the service priorities of the traffic. Energy conservation is emphasized for delay-tolerant traffic, but energy conservation is sacrificed for delay-intolerant traffic. The inability of QoS routing to maintain guaranteed levels of service in an energy-efficient manner for packet radio networks leads us to examine routing protocols that account for the QoS priorities but do not necessarily provide guaranteed levels of service. We exploit the fact that a given path through the network typically provides better service for some classes of traffic than others. By routing different classes of traffic according to the appropriate combinations of metrics, each class uses routes that are compatible with its service priorities, energy is conserved when the service priorities permit, and a more diverse utilization of network resources is achieved (e.g., our multimedia routing protocols are more likely to use multiple paths).

In related work [PRW04c], we have devised and evaluated an adaptive routing protocol that uses a measure of network activity in making routing decisions. Each radio monitors the amount of time that it is transmitting or receiving, and it uses this information to calculate an on-air ratio. This on-air ratio provides each radio with a measure of its activity level, which is used by the radio to determine when to adapt its routing protocol. When the network activity is low, a radio strives to employ low-energy links in forwarding packets. As the activity level increases, the routing protocol adapts by employing more high-quality routes in an effort to maintain low delay and high packet success rate. Our simulation results for mobile networks show that routing based on network activity provides energy savings when the opportunity exists, yet it achieves low delay and high packet success probability.

Highly mobile ad hoc networks for use in environments of interest to the DoD must be able to perform well in situations for which the propagation losses (e.g., due to shadowing) may vary significantly in different regions of the network. In particular, it is possible that packet radio networks, especially those in which the radios have ranges on
the order of a kilometer or more, will experience connectivity patterns in which one or more terminals are within range of a large number of the other terminals. For example, a terminal may be located on a hill, thereby providing line-of-sight connectivity to many terminals in the network. Terminals that experience relatively low propagation losses on the links to a large number of other terminals or for some other reason have a high degree of connectivity with other terminals are referred to as advantaged nodes or advantaged terminals. An advantaged terminal can easily become a bottleneck for the network if the link and network protocols simply select the minimum-energy or minimum-hop routes. We have investigated networks with advantaged terminals in [PRW04c], and we have shown that incorporation of our measure of network activity into the least-resistance routing protocol ensures that the network can achieve high throughput efficiency at low traffic loads and maintain high throughput at higher traffic loads. In particular, the activity metric is a critical component for routing protocols in networks with advantaged terminals, because it is necessary to prevent the network-layer protocols from creating artificial bottlenecks at the advantaged terminals by overloading them with traffic. Our protocol offers the first known solution to this problem.

A significant accomplishment during the MURI project is the development of a new distributed channel-access protocol for broadcast transmission scheduling in mobile ad hoc networks. Broadcast scheduling protocols operate by establishing transmission schedules in which each terminal is assigned transmission times so as to avoid contention for the channel or the intended receivers. In [HaR04] we describe a simple distributed algorithm that permits terminals to schedule collision-free transmissions with the only requirement that a terminal know local information about which terminals are its neighbors and the neighbors of its neighbors. Our primary contribution is described in [AHN04] in which we present and evaluate a distributed and adaptive transmission-scheduling protocol (DATSP) to address the problems of maintaining collision-free transmission schedules due to changes in network connectivity. The DATSP allows each terminal to adapt quickly to each change in the network topology by incrementally updating transmission assignments so as to continually maintain collision-free transmissions. In [WHN04] we describe our new transmission schedule construction protocol (TSCP); this protocol supplements the DATSP by efficiently handling the special situation in which a new terminal joins the network. A journal article based on [WHN04] has been submitted to the *IEEE Transactions on Wireless Communications*. In [WHR05] we describe a new load-based transmission scheduling protocol (LOBATS). The LOBATS is an extension to the DATSP that permits a terminal to increase its effective transmission rate by increasing the number of scheduled transmissions if the terminal has determined it is a local bottleneck.
The significance to the DoD of our new suite of transmission-scheduling protocols is that these protocols permit a mobile ad hoc network to achieve very high levels of utilization of the channel while imposing very low overhead under many conditions. We have developed an analytical upper bound on the maximum stable end-to-end throughput that can be supported in a wireless network that employs broadcast transmission scheduling, and we have shown that the LOBATS can achieve performance close to this upper bound in a wide variety of networks. For example, in a network with 100 terminals and a random traffic generation model in which each terminal generates the same average level of traffic, the LOBATS can achieve network throughput that is more than four times larger than a time-slotted contention-based protocol when the traffic generation rate is very high.

Our transmission scheduling protocols can provide very good network performance at high traffic levels for medium to large networks (i.e., ten to a few hundred terminals) as long as the network density is moderate (i.e., the average number of neighbors is less than approximately 20) and the level of mobility is within the capability of the adaptive protocols. However, our protocols should be seen as an alternative too rather than a replacement for contention-based protocols (e.g., RTS/CTS based protocols). For example, our protocols are most likely to be useful in a network architecture in which some portion of the network must serve in the role of a backbone and must support very heavy levels of traffic for extended periods of time. One implementation approach may be to develop a hybrid protocol that employs our transmission scheduling protocols during periods of high traffic and reverts to a contention-based protocol when the traffic level is low, the density of the network is very high, or the mobility is such that the network connectivity changes more rapidly than can be supported by any topology-dependent transmission assignment protocol.

We considered two control problems for wireless networks in which the terminals are capable of directional transmission and reception. The first problem is scheduling transmissions between nodes so as to maximize the number of traffic flows that can be handled simultaneously according to their desired services. The second problem is accommodating high-priority delay-sensitive flows by forming ephemeral direct links between sources and destinations for those flows, provided that doing so does not result in loss of desired service for any other flow. For both of these two problems, we provide simple distributed algorithms in which each node makes greedy decisions based upon locally-available information measured by the node and received from other nodes.
The link scheduling algorithm, discussed in [Ste03], is designed for unicast and multicast communication and thus complements previous the node scheduling algorithms for broadcast communication. Time is segmented into reserved and contended slots. A slot is available for reservation between a node and one of its neighbors if neither the node nor the neighbor has already reserved that slot or overheard transmissions in that slot from nodes lying in the same direction as the other node. A slot reservation is negotiated by exchanging information about available slots between a node and one (unicast) or more (multicast) of its neighbors. The constraints on slot availability can be generalized to include not only direction but also frequency and code of overheard transmissions. Moreover, the negotiation of transmission opportunities can be applied in floor acquisition as well as in slot reservation. Recently, we have adapted our link-scheduling algorithm for use in multi-frequency networks, providing both slot-reservation and floor-acquisition versions, as discussed in [Ste05].

The link-formation algorithm for delay-sensitive flows computes estimates of delay and cost (e.g., expected number of nodes which may experience interference from the transmissions) for both the multihop path and the direct link between source and destination and selects the route with lower cost, provided the delay constraint for the flow is met on that route. We describe a number of different cost functions in [Ste04]. Accuracy of cost estimates, which depends in part on the particular cost function, has a significant impact on the correctness of the choice of route, as determined through simulation. When establishing a direct link, the source and destination must exchange sufficient information over the existing multihop path to determine when to communicate, in what direction, and at what power.

The heuristic and local nature of each of these two control algorithms does not permit us to make any guarantees about the solutions generated. Specifically, we cannot guarantee that the link scheduling algorithm yields solutions that are optimal or within a given factor of optimal, and we cannot guarantee that the link-formation algorithm yields solutions that do not adversely affect other flows. Nevertheless, we have demonstrated that these algorithms perform well in simulation, in the presence of time-varying network graphs, traffic loads, and traffic patterns, yielding solutions that not only support simultaneous traffic flows with differing service needs but also require only a small number of packet exchanges to do. Thus, these algorithms appear well-suited for deployment in tactical networks in which a wide range of service needs must be satisfied with a low probability of detection and, in the case of battery-powered nodes, a low rate of energy dissipation. These algorithms can be used with beam-switching, beam-steering, or beam-forming antennas, and, in fact, with beams of any width, but we note that narrow
beamwidths and accurate beam pointing significantly reduce the potential for interference.

We have developed a concatenated coding scheme that provides a significant performance improvement over traditional turbo codes in many situations [She01a]. In particular, these codes have a much lower error floor than turbo codes. The codes are based on turbo and simple parity-check codes, and do not have a significantly higher decoding complexity than turbo codes alone. We generalized these codes to use multidimensional product codes and further studied the properties of multidimensional product codes in [ShW03].

We also developed a new incremental-redundancy hybrid ARQ scheme that retransmits the bits that are deemed unreliable by soft-output decoding algorithms. This reliability-based hybrid ARQ (RB-HARQ) scheme can provide performance approaching the capacity for AWGN channels [She02]. RB-HARQ techniques are particularly appropriate for bursty channels, such as for channels with time-selective fading or partial-time jamming. We investigated RB-HARQ schemes for partial-time jamming channels in [RMS05].

We investigated approaches to improve link and end-to-end throughput in ad hoc networks by taking advantage of capability differences between neighbors of a transmitter. We apply simulcasting in the physical layer, in which unequal error protection signaling is used to send additional message streams at very little performance degradation to the other nodes in the system. In [JuS04] we investigated the use of simulcasting in ad hoc networks. We consider both physical-layer signal design and modifications to the higher-layer protocols to effectively utilize the capability provided by simulcasting to enhance network performance. The performance was investigated through analysis and simulation. The results show that simulcasting can significantly improve the link and end-to-end throughputs while having minimal impact on other performance metrics, such as network connectivity.

We have further developed our cooperative reception techniques based on distributed decoding and signal processing and investigated the application of these techniques to channels with jamming and fading. The techniques that we have developed achieve spatial diversity through cooperation among a group of receivers. The group of receivers effectively forms a distributed antenna array that can function efficiently without physical connections among the receivers. An overview of cooperative communication techniques is given in [ASW04a]. In [MSW04b] we consider the design of cooperative
reception and signal processing techniques for communication in the presence of partial-
time jamming. We show that the usual approaches that are based on maximal-ratio 
combining are not effective for these channels, and we develop distributed beamforming 
and interference-cancellation techniques that are effective at mitigating the effects of the 
jamming. The techniques that we have developed can also be used on channels with both 
partial-time jamming and quasi-static fading. However, the difficult problem in these 
channels is the estimation of the unknown signal amplitude and phase in the presence of 
the jamming signal. We have developed effective estimators for these channels, and 
pilot-assisted and blind channel estimation techniques based on the expectation-
maximization (EM) algorithm are presented in [MWS05].

In [JoK03], systems using $M$-ary orthogonal signaling and $Q$-ary RS or BCH error 
control codes were presented where $M<Q$ for a fixed overall system bandwidth.
Analytical expressions, which include all possible code symbol groupings of channel 
symbols, were developed to compare the bit error probability performance of comparable 
systems. These results were based on individual codewords using errors-only decoding 
and errors and erasures decoding with transmission over a Rayleigh fading channel.
Results showed that as the energy-to-noise density ratio was increased, reduced signal 
systems outperformed systems with larger signal constellation sizes, in terms of bit error 
probability, due to the improved error performance of the codes.

In [KoJ04], a simple and easy to evaluate series expression for evaluating the channel 
capacity of the binary input AWGN channel was developed, which precludes the 
necessity of numerical integration. In addition, tight upper and lower bounds were 
developed. These bounds also provided good approximations for the capacity and are 
within 0.1% relative error (within 0.01% for upper bound) for $C\geq0.457$ and a 
corresponding $E_b/N_0\geq0$ dB.

We introduced a new approach to multivariate polynomial factorization [AGL04] that 
incorporates ideas from polyhedral geometry, and generalizes Hensel lifting. The main 
contribution is to present an algorithm for factoring bivariate polynomials which is able 
to exploit to some extent the sparsity of polynomials. The algorithm is implemented to 
factor randomly chosen sparse and composite polynomials of high degree over the binary 
field. In [BGL03], we first give an exact formula for the probability that a random 
Krylov subspace is equal to whole vector space over a finite field, then we derive a 
simple explicit bound for this probability. These results are useful for the analysis of 
many parallel (or block) algorithms for solving linear equations over finite fields.
We developed a method to find general multivariate Pade approximation using the Grobner basis technique [FaG06]. This method is more flexible than previous approaches, and several examples are given to illustrate its advantages. When the number of variables is small compared to the degree of approximation, the Grobner basis technique is more efficient than the linear algebra methods in the literature. In [FaG05] we show how the Grobner basis techniques can be used in coding theory, especially in the construction and decoding of linear codes. A new method is given for construction of a large class of linear codes. These codes include as special cases many well-known codes such as Reed-Solomon codes, Hermitian codes and, more generally, all one-point algebraic geometry codes. One major advantage of these codes is that there is an efficient algorithm for errors-and-erasures decoding, which is important for wireless communication networks in which side information can be derived (e.g., [GPR01], [MaP02b], and [PuS03a]). Another major advantage is that they can be defined on random points in any affine space, so it may be useful in other applications such as gene networks in biology where microarray data inevitably contain noise. The research on Hermitian coding illustrates the advantages of the interactions among investigators that are possible in such a multidisciplinary project. There was a strong coupling between research on decoding algorithms for Hermitian codes and the applications of Hermitian coding to FH spread-spectrum networks that resulted in [MaP03].

A new algorithm is developed in [Gao03a] for decoding Reed-Solomon codes. It uses fast Fourier transforms and computes the message symbols directly without explicitly finding error locations or error magnitudes. In the decoding radius (up to half of the minimum distance), the new method is easily adapted for error and erasure decoding. It can also detect all errors outside the decoding radius. Compared with the Berlekamp-Massey algorithm, discovered in the late 1960's, the new method is simpler and more natural, yet it has a similar time complexity. In [Gao03b], we presented for factorization of bivariate polynomials over any field of characteristic zero or of relatively large characteristic. It is based on a simple partial differential equation that gives a system of linear equations. The new algorithm finds absolute and rational factorizations simultaneously and is easy to implement for finite fields, local fields, number fields, and the field of complex numbers. The theory of the new method allows an effective Hilbert irreducibility theorem, thus an efficient reduction of polynomials from multivariate to bivariate.

A numerical algorithm is presented in [GKM04] for finding approximate factorization of multivariate polynomials whose coefficients are given approximately as floating-point numbers. It is based on the algorithm in [Gao03b] and makes repeated use of singular value decompositions. A significant body of experimental data shows that this algorithm
is practical and can find factorable polynomials within a distance that is about the same in relative magnitude as the input error, even when the relative error in the input is substantial.

A polynomial time algorithm is given in [GaN05] for the computation of the fault tolerance of any given Cayley graph over any group. A new structural result for Cayley graphs is proved and this result yields simple proofs of optimal fault tolerance for two infinite classes of Cayley graphs, namely exchange graphs and quasiminimal graphs.

Ostrowski established in 1919 that an absolutely irreducible integral polynomial remains absolutely irreducible modulo all sufficiently large prime numbers. In [GaR03], a new lower bound for the size of such primes is given in terms of the number of integral points in the Newton polytope of the polynomial, significantly improving previous estimates for sparse polynomials. In [GRS03], we study the intrinsic relationship between a Grobner basis and the algebraic variety it defines. It is shown how a minimal Grobner basis reveals the geometric structure of the variety, and vice versa. These relationships can be used to decompose polynomial systems into smaller systems, hence useful for solving polynomial systems.

One phase of the research on error-control coding was devoted to codes from graphs and designs with a view to using their combinatorial properties to develop techniques that will assist in the practical use of the codes. For example, decoding algorithms using the permutation group of the code, analysis of the weight enumerator of the codes, finding bases of minimum-weight vectors, minimum weight of some geometric codes, are some of the areas that have been explored and have lead to useful results, all of which apply to infinite classes of codes.

In [CHK03], [CKR02], and [KeR03] the minimum weight of some classes of geometric codes were determined; these codes are related to generalized Reed-Muller subfield subcodes, and have good parameters. Their well-known geometric properties are used to deduce properties of the codes. Similarly, in [DiK00] and [DiK02], bases of minimum weight vectors of generalized Reed-Muller codes and projective generalized Reed-Muller were found, giving sparse matrices as generators for the codes. In [KMR04d] properties of the binary codes arising from symplectic geometry were established.

Permutation decoding, first developed in the 1960's, is a useful and efficient decoding tool when the code has a large automorphism group. The codes from graphs and designs satisfy this condition and PD-sets for using the full error-correction capability of the code for a number of classes were discovered: in [KMR04a] these sets were found for binary
codes of triangular graphs, and in [KMR04b] and [KMR04c], properties of the binary codes of graphs from triples were found, leading to PD-sets for some of the better classes of codes from these graphs. Square and rectangular lattice graphs were studied for the same purpose in [KeS04] and [KeS05], and full PD-sets found for all classes.

The prospect of finding PD-sets for all parameters of classes of codes arising from geometries was shown to be unattainable for the codes from finite geometries due to the size of the automorphism groups not increasing fast enough with the order of the geometry to match the minimum weight. For these, partial permutation decoding was used, by finding so-called s-PD-sets to correct s errors, where s might be less than the full potential of the code. This work appears in [KMM04a] for finite planes, and in [KMM04b] for higher dimension; the latter paper also develops information sets for the generalized Reed-Muller codes. A similar examination for partial permutation decoding was made in [KeL04], where quadratic residue codes and other codes from Paley graphs were examined for s-PD-sets. The s-PD-sets in these cases are very small, giving a low worst-case time complexity for this method in this case.

Antenna performance characteristics can be controlled by various forms of loading, which enable one to increase bandwidth of input VSWR and maintain gain quality over a reasonably wide range of frequencies. The analysis of an antenna loaded with lumped circuit elements typically is based on an efficient combination of Maxwell's equations and circuit theory. If for example a wire antenna is loaded by a traditional thin-wire coil, the accuracy of the analysis may suffer due to the field coupling between the antenna members and the turns of the coil, which is not accounted for in either the antenna or circuit analysis per se. In such cases, the advantage in simplicity gained by using circuit theory to account for the presence of the lumped loading circuit elements is diminished by a loss in accuracy of the combined analysis. Shielding a coil, however, isolates its windings from the "stray fields," thereby ensuring coupling only at the terminals of the loading circuit and assuring the validity of computations based on the laws of circuit theory. Of course, there are other advantages realized by shielding tuning coils. The use of circuit laws to characterize load circuits, as opposed to appealing to Maxwell's equations to capture all significant effects, greatly simplifies analyses and reduces design efforts. On the other hand, the presence of the shield renders the characterization of a coil more complex. However, if the input impedance to a shielded load can be determined accurately, one may easily employ such loads for tuning purposes.

An accurate hybrid method has been developed for analyzing a cylindrical antenna loaded with a shielded coil or with multiple shielded coils. The method consists of a
computational procedure for solving the loaded cylindrical antenna integral equation and of a measurement procedure for accurately characterizing the shielded load. The structure of the shield ensures that the above mentioned accuracy-degrading effects of stray field coupling are reduced to nearly zero and the hybrid analysis technique fully accounts for the loading coil in the presence of its shield. Measuring the input impedance of a shielded load presents several difficulties in practice, as standard connectors do not exist for interfacing the load with a network analyzer.

A method to accurately measure the impedance looking into a shielded load, which involves interfacing the load with a measuring instrument by means of a two-port network, is described. The effects of the interfacing network on the measurements of the properties of a shielded load are removed from the data by invoking a simple transformation based upon two-port network theory and knowledge of the impedance parameters of the network. A calibration scheme was developed to determine the impedance parameters of the interfacing network. With data available to characterize the shielded load, one incorporates this load in the antenna integral equation and can obtain a solution that accurately accounts for the presence of loads in their shields along the antenna. Data from the hybrid solution are compared with those obtained from measurements made on a laboratory model. Accurate measurement of the input impedance of the model is facilitated by another de-embedding technique similar to that mentioned previously.

Throughout the course of this investigation of broadband, omni-directional antennas several techniques for enhancing the bandwidth of cylindrical monopoles and dipoles have been developed. Along with the study of methods for extending bandwidth while retaining desirable features of simple cylindrical radiating structures, we have developed analytical and numerical methods which enable us to combine, under the control of optimization routines, several bandwidth enhancement schemes in a single radiator. Initially we increased operating bandwidth by "tuning" cylindrical antennas with RLC networks placed inside the conducting tubes (from which the radiator is fabricated) as described in [BuR02] and [PBR04], combined with appropriately designed matching networks [RBM03]. Component values and configurations of the matching networks and tuning circuits are arrived at by optimization methods [BRM01] coupled to integral equation analyses [RoB01] of the composite antennas. The tuning circuits have taken on numerous configurations as we learned how to squeeze more bandwidth and performance from a structure [LBR04a]. More recently [LBR04b], for tuning purposes we have employed cavities interior to the cylindrical antennas that are coupled to the currents on the structure via slots in the antenna's tubular walls. The geometry, size, material inside
the cavities, and the locations of the coupling slots afford a wide range of parameters with respect to which one can perform optimization analyses to arrive at desired antenna performance. All of our theoretical predictions have been corroborated by measurements on laboratory models and, in some important cases, values of parameters in integral equations have been determined from laboratory measurements.
MURI PUBLICATIONS


[KMM04b] J. D. Key, T. P. McDonough, and V. C. Mavron, "Information sets and partial permutation decoding for codes from finite geometries", accepted for publication in Finite Fields and their Applications.


[KMR04c] J. D. Key, J. Moori, and B. G. Rodrigues, “Binary codes from graphs on triples and permutation decoding”, accepted for publication in *Ars Combinatoria*


[KeS04] J. D. Key and P. Seneviratne, “Permutation decoding of the binary codes of lattice graphs,” accepted for publication in *Discrete Mathematics*.


of the 2004 IEEE Military Communications Conference (Monterey, CA), November 2004.


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Faculty and Student Honors and Awards

Faculty Honors and Awards:

Three IEEE Millennium Medals, 2000
National Science Foundation CAREER Award, 2001
Three Clemson University Board of Trustees Awards for Faculty Excellence, 2001
Clemson University Board of Trustees Award for Faculty Excellence, 2002
Clemson University Board of Trustees Award for Faculty Excellence, 2003
Clemson University Board of Trustees Award for Faculty Excellence, 2004
Clemson University Alumni Award for Outstanding Achievement in Research, 2000
Clemson University Alumni Award for Outstanding Achievement in Research, 2001
Honorary Member, Golden Key National Honor Society, 2000
IEEE Communications Society Distinguished Lecturer, 2001-06
Edwin Howard Armstrong Achievement Award, IEEE Communications Society, 2003
Finalist Eta Kappa Nu Outstanding Young Electrical Engineer, 2004
University of Southern California Viterbi School of Engineering Distinguished Alumni Award, 2005

Graduate Student Awards:

AFCEA Fellowship, 2001
Clemson College of Engineering and Science Outstanding Graduate Researcher, 2001
Clemson University Outstanding Graduate Researcher, 2001
Two MIT Lincoln Laboratory Fellowships, 2002-05
National Science Fellowship, 2004-08

Undergraduate Student Awards:

IEEE Region 3 Student Paper Competition, First Place, 2001