Reliable Communication using Packet Coding for Underwater Acoustic Channels

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***Abstract*—We investigate reliable data packet delivery employing random linear packet coding for a half-duplex underwater acoustic link. Packet coding is performed on a group of M information-bearing packets so as to form $N \geq M$ coded packets, where N is chosen such that a pre-specified probability of successful decoding is maintained at the receiver. We regard a group of N coded packets as one super-packet, on which we apply an ARQ technique. Specifically, we group L super-packets and apply a group stop-and-wait acknowledgment procedure to the so-obtained unit (a super-group) to achieve full reliability. We also explore adaptive power and rate control combined with the proposed technique in order to improve the performance on fading channels. Using the channel gain information obtained via feedback from the receiver, the transmitter adjusts its power and coding rate such that the average energy per bit is minimized. In doing so, two constraints are observed, one on the maximum transmit power and another on the size N of the super-packet. Under these constraints, we develop the adaptation policy and provide analytical results for the throughput efficiency. Finally, we compare the performance of the proposed technique to that of conventional stop-and-wait, as well as a full-duplex benchmark, showing that packet coding technique on a half-duplex link can achieve a throughput efficiency that is very close to that of a full-duplex link.**

I. Introduction

Acoustic communication has found applications in a variety of systems such as those involving multiple underwater autonomous vehicles, deep-sea oil and gas field maintenance, climate recording and biological eco-system monitoring. Acoustic signals propagate well underwater, but bring a number of challenges for communication. Since the speed of sound is low (nominally1500 m/s), acoustic communication suffers from long propagation delays which contribute to link latency, and pronounced Doppler distortion which contributes to high packet loss rates. Combined with the half-duplex nature of most acoustic modems, these challenges make traditional techniques for reliable communication, developed for terrestrial links, inefficient in underwater systems.

In this paper, we address the issue of reliable packet delivery for an underwater acoustic link. Traditionally, automatic repeat request (ARQ) protocols such as stop-and-wait (S&W), go-back-N, and selective repeat are used to make a link reliable. Because of the long propagation delays experienced by the acoustic links, these techniques become inefficient as they rely on a timely feedback from the receiver. As most acoustic modems operate in half-duplex mode, the choice of a data link layer protocol is further limited to the S&W type. The performance of ARQ techniques for underwater acoustic channels was discussed in [1]. Two modifications of the basic S&W techniques were studied there, and their performances were compared in terms of the throughput efficiency. It was shown that grouping of packets improves the throughput efficiency. Reliable data transfer from one half-duplex node to another was addressed in [2], where the long propagation delay was exploited by allowing the two nodes to transmit simultaneously in a juggling fashion.

In this paper, we explore random linear packet coding as an attractive addition for achieving link reliability. In a packet coded system, a group of M information-bearing packets are encoded into $N \geq M$ coded packets for transmission. The receiver can decode the original information bearing packets from a subset of *any* M out of the N coded packets. It should be noted that the packet coding does not replace channel coding, but can be used in addition to it. Since packet coding is applied at the packet level (as opposed to bit level), this technique can be readily applied to most commercially available acoustic modems.

Packet coding for underwater channels was studied in [3–5]. In [3], rateless codes were considered for reliable data transfer in underwater acoustic networks. It was shown there that the throughput efficiency improved since the feedback was used less often. In [4], the authors investigated optimal schedules for packet coding in a half-duplex link and showed that an optimal number of coded packets exists, which minimizes the time (or energy) required to complete the transmission of a group of packets. Optimal broadcasting strategies for broadcasting information using random linear packet coding were addressed in [5], showing performance improvements over traditional ARQ techniques.

Random linear packet coding for acoustic links was considered in [6]. It was shown that a small additional redundancy suffices to maintain a pre-specified reliability at a relatively high level at the receiver. In this paper, we extend the work of [6] to design a fully reliable link using packet coding in conjunction with an ARQ mechanism. In particular, we regard a group of N coded packets as one unit called a super-packet. We group L such super-packets together, forming a super-group which is then transmitted. The receiver sends a selective acknowledgment for the super-group of L units. If a certain super-packet is negatively acknowledged, it is re-transmitted in the next round, along with any new units. Packet grouping hierarchy is illustrated in Fig. 1.

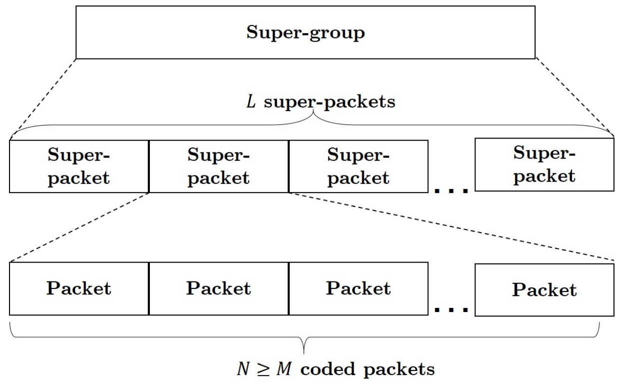

Figure 1. Grouping hierarchy of the proposed packet coding technique.

We also explore the use of adaptive power and rate control to further improve the performance on fading channels. Adaptive power and rate control for an underwater link employing random linear packet coding were investigated in [6]. We use the same adaptation policies as in [6] to minimize the average energy per bit while also achieving full reliability by employing the super-packet ARQ. Two constraints are imposed in the optimization; the transmit power cannot exceed a maximum available level, and the number of coded packets cannot exceed a maximum value, dictated by the coherence time of the channel, the maximum permissible decoding delay and the minimum acceptable bit rate.

We compare the proposed technique with a full-duplex link (as a benchmark) and other techniques employing packet coding, in terms of the throughput efficiency. We show that the proposed technique achieves throughput efficiency which is very close to that of the full-duplex link. The rest of the paper is organized as follows: We present the system model in Sec. II. Sec. III is devoted to power and rate control. The results are presented in Sec. IV, and the conclusions are summarized in Sec. V.

II. System Model

A. Packet Coding

The transmitter in a packet coded system buffers a group of M information-bearing packets, and encodes them into $N \geq M$ coded packets. Each packet contains N_b bits. It has to be noted that the packet coding does not replace channel coding, but works to improve performance in addition to channel coding. The packet duration is given by $T_p = N_b/R_b$, where R_b is the bit rate in the channel.

The N coded packets are transmitted over a channel which is characterized by the packet error rate P_E. For simplicity, we assume differentially coherent detection with no coding or diversity. We note that our analysis does not change with the change in any of these assumptions; only numerical results do. The packet error rate is related to the bit error rate P_e as $P_E = 1 - (1 - P_e)^{N_b}$. The receiver can decode the original information-bearing packets as soon as it receives *any* M coded packets. A packet is deemed successfully received if it passes a standard procedure such as cyclic redundancy check (CRC).

We define the success rate or the reliability P_s as the probability that at least M out of the N coded packets are received correctly, i.e.,

$$P_s = \sum_{m=M}^{N} \binom{N}{m} (1 - P_E)^m P_E^{N-m} \tag{1}$$

The optimal number of coded packets N_M to transmit in each group is chosen such that a pre-defined probability of success P_s^* is maintained at the receiver. In other words, N_M is the smallest N for which $P_s \geq P_s^*$.

B. ARQ

In [6], we analyzed packet coding that is designed to achieve a pre-specified success rate. When used without power / rate control, this design yields a system that needs no feedback, and as such is appealing for half-duplex acoustic systems with long delay. If power / rate control is used, feedback is needed to inform the transmitter of the channel state. That feedback can be used sparingly if the large-scale channel variation is slow. Namely, if the channel is deemed to stay more or less constant over several minutes, feedback is needed only as often.

The feedback that we considered in [6] was used only for conveying the large-scale, slowly changing channel gain, and not for ARQ. As a result, the scheme guaranteed only a pre-specified reliability, which is less than 100%. To achieve full reliability, an ARQ procedure must be used. Traditional ARQ methods for the underwater channel are based on the variants of the Stop-and-Wait (S&W) protocol [1]. The first of these variants, S&W-1 protocol, transmits packets one by one, waiting for an acknowledgment between each packet. The throughput efficiency of the S&W-1 protocol is given by

$$\eta_{SW1} = (1 - P_E)\frac{T_p}{T_p + T_w} \tag{2}$$

where T_w is the waiting time, which is at least equal to the round-trip propagation delay plus any synchronization/acknowledgment overhead.

To overcome the throughput limitation of the conventional S&W-1 protocol, packet grouping can be used with selective acknowledgments. The resulting S&W-2 and S&W-3 protocols have greater throughput efficiency, particularly on channels with long delay. Specifically, S&W-2 transmits groups of K packets in each round before waiting for an acknowledgment. If a packet needs to be re-transmitted, it is included into the next group of K packets that now contain both new and old packets. In contrast, S&W-3 transmits each group of K packets until it is successfully received, i.e. no new packets are added when re-transmission is made. Since S&W-2 has better throughput performance than S&W-3 [1], we only consider S&W-2 in the present analysis. The throughput efficiency of the S&W-2 protocol is given by

$$\eta_{SW2} = (1 - P_E)\frac{KT_p}{KT_p + T_w} \tag{3}$$

To extend the S&W-2 principle to a packet-coded system we regard N_M coded packets (obtained from M information packets) as one unit called a super-packet. We group L such units together to form a super-group. The receiver sends a selective acknowledgment for the super-group. If a certain super-packet is negatively acknowledged, it is re-transmitted in the next round (next super-group), along with any new super-packets. This approach is somewhat wasteful, in the sense that if a super-packet is negatively acknowledged, one does not have to re-transmit all of its N_M packets. It suffices to re-transmit only so many packets as there are missing degrees of freedom (extra coded packets needed to complete the decoding of the original M). However, the chances of a re-transmission being needed are contingent upon the *pre-defined* success rate P_s^*, which is a design parameter and can be set high. In other words, if $P_s^* = 0.99$, only one in every 100 super-packets will need to be re-transmitted on average. The resulting re-transmissions are not overly wasteful, especially in light of the procedure's complexity. Our procedure is rather simple, as it does not require the missing degrees of freedom to be spelled out explicitly.

Although analogous to S&W-2, the proposed procedure differs in the fact that each unit is not one information packet, but one super-packet. Hence, the role played in the original S&W-2 by the packet error rate P_E is now played by $1 - P_s^*$. The success rate P_s^* depends on P_E, but packet coding ensures that $1 - P_s^* \ll P_E$. In other words, P_s^* can be controlled through packet coding, and we use this fact to improve the throughput efficiency. Our procedure also differs from the one considered in [4], where a single unit of $N \geq M$ coded packets are transmitted until all its M information-bearing packets have been successfully received. Such an operation is accomplished by providing feedback on the missing degrees of freedom (which diminish with repeated feedback).

The throughput efficiency of the proposed packet coding based technique is given by

$$\eta_{pc} = P_s^* \frac{LMT_p}{LN_MT_p + T_w} \tag{4}$$

Fig. 2 shows the throughput efficiency as a function of the super-group size L. Clearly, increasing L leads to increased throughput efficiency. However, it should be noted that a greater L increases the decoding delay at the receiver, and also requires larger buffers to store all the packets at both the transmitter and receiver. A practical choice L might thus be limited by the buffer available at the transmitter and receiver, and the maximum permissible decoding delay.

Fig. 3 shows the throughput efficiency as a function of the group size M. Again, larger M leads to higher throughput efficiency but also requires a larger buffer to store packets at the transmitter and receiver. As a result, the choice of M is also dictated by the buffer size at the transmitter and receiver.

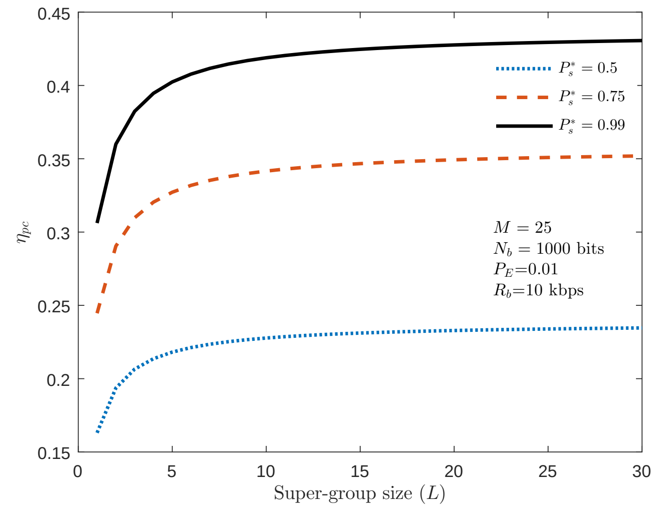

Figure 2. Throughput efficiency η as a function of the super-group size L.

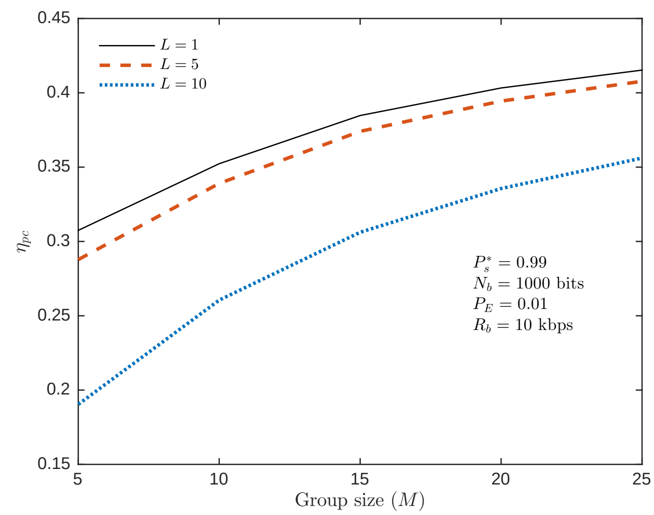

Figure 3. Throughput efficiency η as a function of the reliability P_s^*.

Finally, Fig. 4 shows the throughput efficiency as a function of the success rate P_s^* for different values of the super-group size L. As expected, the throughput efficiency increases with higher P_s^*.

III. Adaptive Power and Rate Control

Our analysis so far focused on a time-invariant channel with a fixed packet error rate. On a time varying channel, the bit error rate P_e is a function of the signal-to-noise ratio (SNR), which is in-turn a function of the channel gain G. The SNR is related to the transmit power P_T and the noise power P_N as $\gamma = GP_T/P_N$. The probability of packet error is now a function of γ, and is related to the bit error rate as $P_E(\gamma) = 1 - (1 - P_e(\gamma))^{N_b}$. Since the channel gain G is

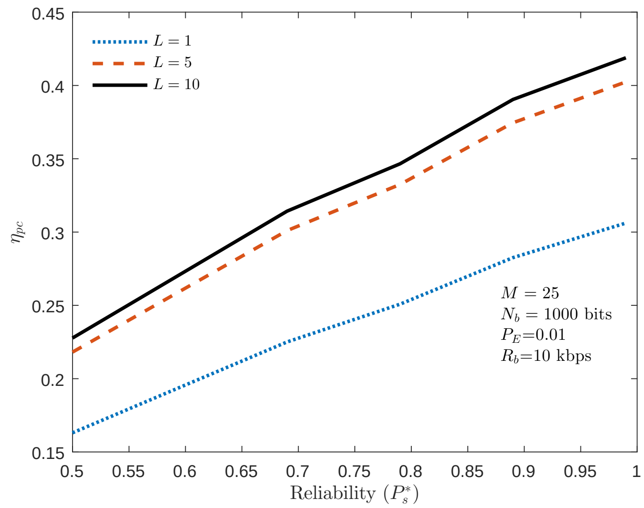

Figure 4. Throughput efficiency η as a function of the reliability P_s^*.

randomly varying, so are the packet error rate $P_E(\gamma)$ and the probability of successful decoding $P_s(\gamma)$.

Adaptive power and rate control for an acoustic link employing packet coding were analyzed in [6]. In this section, we extend the work of [6] to a link which provides full reliability. In particular, we aim to determine the transmit power P_T and the number of coded packets N so that the reliability $P_s(\gamma)$ is maintained at a pre-defined level P_s^*, while also minimizing the average energy per bit,

$$\bar{E}_b = \frac{1}{R_b}\frac{E\{NP_T\}}{P_s^* M} \tag{5}$$

The outage probability P_{out} is defined as the probability that $P_s(\gamma) \leq P_s^*$. The outage gain corresponding to a given P_{out} is given by G_{out}.

We impose two main constraints: (a) the transmit power is limited to $P_{T,max}$, and (b) the maximum number of coded packets is limited to N_{max}. The value of $P_{T,max}$ is dictated by the total power budget or by the system, while the value of N_{max} is determined such that a group does not exceed the coherence time of the channel, maximum decoding delay, or a minimum tolerable average bit rate.

We use the same adaptation policy outlined in [6] but extend it to work along with an ARQ technique to provide full reliability. The channel gain information is sent along with the acknowledgment every L super-packets. The choice of L is now also limited by the coherence time of the channel along with the maximum permissible decoding delay and the buffer size at the transmitter and receiver. The time taken to send the L super-packets must not be more than the coherence time of the channel because a feedback is necessary to update the channel gain.

IV. Results

In this section we compare the performance of our proposed packet coding technique in terms of the throughput efficiency. For comparison, we use the full-duplex (FD) link as the benchmark technique. For a full-duplex link, we assume that the transmitter and receiver are capable of transmitting/receiving simultaneously. The transmitter in this case sends packets back-to-back until an ACK is received from the receiver indicating that decoding was successful. The average time taken to complete the transmission of a group of M packets on a full-duplex link is given by

$$T_{FD} = \frac{MT_p}{1 - P_E(\gamma)} + T_w \tag{6}$$

and the corresponding throughput efficiency is given by

$$\eta_{FD} = \frac{MT_p}{T_{FD}} \tag{7}$$

We also compare our proposed technique with the optimal technique for a half-duplex link (HD,opt) proposed in [4], where the authors showed that an optimal number of coded packets can be obtained to minimize the average time needed to transmit a group of M packets. The throughput efficiency for optimal half-duplex technique is given by [4]

$$\eta_{HD,opt} = \frac{MT_p}{T_M} \tag{8}$$

where T_M is the average time required to transmit a group of M packets. The closed form solution for T_M is provided in [4].

We assume that the benchmark techniques used for comparison use the same maximum power $P_{T,max}$ as the proposed packet coding technique. For a given channel gain G, the transmitter adapts its transmit power as

$$P_T = \begin{cases} P_{T,max}G_{out}P_N/G, & \gamma^* P_N/G \leq P_{T,max} \\ 0, & \text{otherwise} \end{cases} \tag{9}$$

Fig. 5 show the throughput efficiency of the different methods as a function of the packet size N_b for $L = 5$ (chosen so as to conform to the coherence time of the channel $T_c = 15$ s). For numerical evaluation, we assume Rician fading, differentially coherent detection with no coding or diversity, and that the channel gain G is log-normally distributed [7], i.e., $10\log_{10} G \sim \mathcal{N}(\bar{g}, \sigma_g^2)$. It can be seen that our proposed packet coding technique can achieve efficiency that is comparable to that of the optimal half-duplex (HD,opt) technique but with much lower complexity.

Fig. 6 shows the throughput efficiency as a function of the group size M. It shows that larger group sizes are favored, as long as the group size M and the super-group size L together do not violate the coherence time of the channel. The choice of M and L also depends on the acceptable coding delay at the receiver.

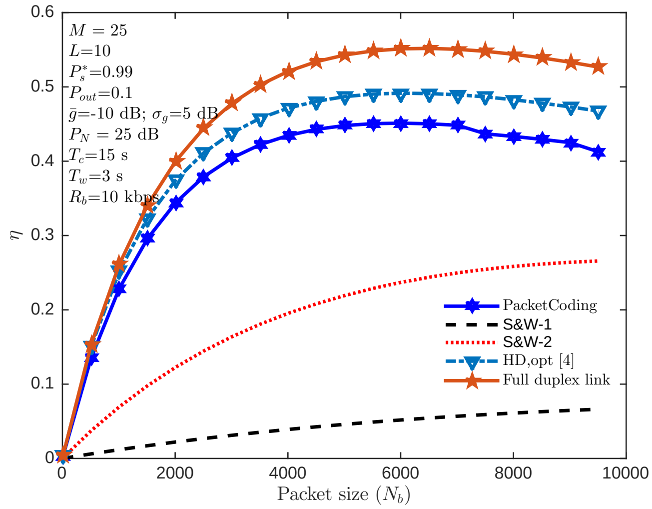

Figure 5. Throughput efficiency η as a function of the packet size N_b.

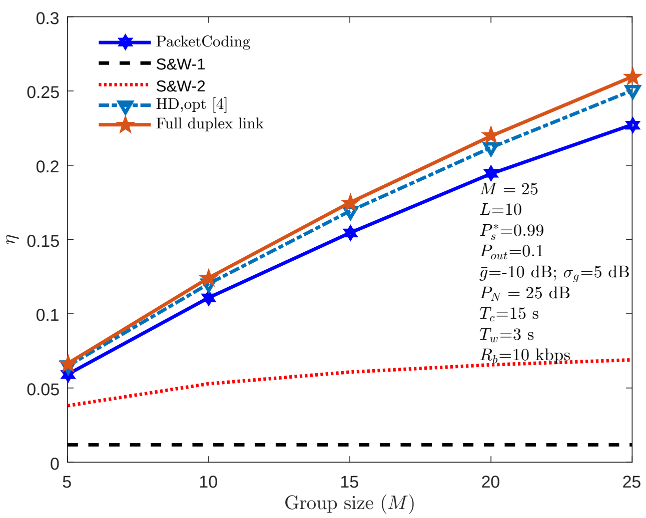

Figure 6. Throughput efficiency η as a function of the group size M under the assumption of log-normal distribution for the channel gain G.

V. Conclusion

Random linear packet coding for reliable data transfer was considered for underwater acoustic links characterized by long delay and large-scale fading that obeys a log-normal model. The number of coded packets to transmit in each group was determined so as to satisfy a pre-defined reliability at the receiver. Each group of packets was considered a super-packet and L such super-packets were combined to form a super-group. A stop-and-wait ARQ technique was employed on the super-group level to achieve full reliability.

Joint power and rate control was employed to overcome the effects of channel fading. Under the assumption of Rician fading, and log-normal distribution for the large scale channel gain, the throughput efficiency of the proposed technique was compared with the full-duplex link, optimal half-duplex technique, and S&W ARQ techniques. It was shown that our proposed low-complexity packet coding technique can achieve throughput efficiency that was significantly better than that of the S&W ARQ techniques, while also comparable to the throughput of the full-duplex and optimal half-duplex link.

For future work, we will concentrate on extending the proposed technique to a broadcast network, where the transmitter will adapt its transmit power and rate based on the mean/minimum of the channel gains received from each node in the network.

Acknowledgments

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