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A Hybrid ARQ scheme combining erasure codes and selective retransmissions for reliable data transfer in underwater acoustic sensor networks

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Abstract

In this paper, we investigate a hybrid automatic repeat request (ARQ) scheme for reliable data transfer in multi-hop underwater acoustic sensor networks (UASNs). The proposed scheme combines Reed-Solomon-based packet level erasure coding and selective retransmissions to achieve high reliability and energy efficiency. We also describe how to select the code rate (i.e., number of encoded packets) for a given message block consisting of *m* packets. The paper evaluates the performance of the proposed scheme by taking into account the underwater-specific characteristics such as distance dependent bandwidth, acoustic spreading effects, propagation loss, and fading effects. Simulation results demonstrate that the proposed scheme can significantly improve the network throughput and energy efficiency while reducing the end to end delay considerably.

Keywords: Hybrid ARQ, Packet level erasure coding, Reliable data transfer, Selective retransmissions, Underwater acoustic sensor networks

1 Introduction

Underwater acoustic sensor networks (UASNs) consist of a number of different types of sensor nodes that are interconnected to one or more underwater sinks by means of wireless acoustic links. Underwater sink nodes are responsible for collecting data from the sensor nodes and forwarding it to the surface sinks, which are equipped with radio frequency (RF) links to the on-shore base stations. UASNs are poised to enable a wide range of underwater-specific applications such as oceanographic data collection, pollution monitoring, disaster prevention, assisted navigation, marine surveillance, etc. [1]. However, research community faces several challenges in the design and development of UASNs. Some of the inherent issues are very high propagation delay as compared to RF transmission, high packet error rate and low available bandwidth that depends on propagation distance. Data transmission reliability is an essential requirement for some of the mission critical applications supported by UASN involving mobile submarines that cooperate to track moving objects. Further, energy efficiency is also a major concern in the design of UASNs, since underwater acoustic sensor nodes are typically powered by batteries, which are difficult to replace or recharge in aquatic environments. Thus, energy efficiency and reliability form two critical design constraints for UASNs [1–5].

Generally, forward error correction (FEC) and packet retransmission-based techniques are used to improve the data transfer reliability in terrestrial wireless sensor networks [6]. FEC techniques employ addition of redundant data to improve the data reliability. Bit level FEC improves the reliability at the physical layer by introducing redundancy at the bit level, and thus deals with the bit errors introduced by the channel. However, it is often not practical to select an FEC scheme that can correct all possible errors. Hence, efficient schemes must be implemented at the higher layers of the protocol stack to deal with uncorrectable errors at the physical layer that quite often leads to packet erasures and losses. Packet level redundancy based schemes such as erasure codes deal with packet

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erasures and losses by introducing inter packet redundancy [7, 8]. In the case of erasure codes, m packets from the source node are encoded to generate $m + k$ encoded packets in such a way that any subset of m encoded packets are just sufficient to recover the original data at the receiver [9]. However, pure FEC based schemes can result in significant number of redundant packet transmissions, thereby affecting the energy efficiency [10]. Automatic repeat request (ARQ) and its deviants rely on retransmissions of corrupted or lost packets upon time outs or after receiving explicit requests from the receiver. Since underwater acoustic channels are characterized by high propagation delay and error rate, ARQ schemes suffer higher end-to-end delay and retransmission-based overheads [4].

Hybrid ARQ schemes, which combine FEC with hop-by-hop retransmission upon failure, are promising candidates for ensuring reliability in UASNs. In this paper, we propose and investigate erasure codes based hybrid ARQ scheme for reliable data transfer in UASNs [4]. The proposed scheme combines Reed-Solomon (RS) erasure codes based FEC with hop-by-hop selective retransmission scheme. Erasure codes form a class of packet level redundancy based FEC scheme that can be used to deal with packet erasures and losses in the network. In this case, m packets from a source are encoded to generate $m + k$ encoded packets in such a way that any subset of m encoded packets are just enough to reconstruct the original data, thus allowing the receiver to recover from k packet losses in a group of $(m + k)$ encoded packets [9]. If more than k packets are lost, the proposed scheme makes use of a hop-by-hop selective retransmission scheme for ensuring reliability. With the help of NACK packets, the receiver of each hop notifies the corresponding sender, the number of data packets missing at the receiver and therefore the sender can decide how many packets need to be retransmitted for successful decoding. We use systematic RS codes as erasure codes with simplified encoding and decoding procedures. The rest of this paper is organized as follows. In Section 2, we provide the literature survey. Section 3 gives an introduction on RS based erasure codes. Section 4 describes the proposed protocol. Section 5 describes the analytical models for evaluating the performance of the proposed scheme. The simulation results are described in Section 6. Finally, the paper is concluded in Section 7.

2 Related work

Recently, extensive research has been reported on the design of protocols for reliable and energy-efficient data transfer in UASNs. As mentioned before, traditional ARQ based schemes are not suitable for UASNs, since sender relies on acknowledgments to decide whether to retransmit the packet or not, leading to an increase in end-to-end delay [4]. Even though the improved versions of ARQ proposed in [11–14] and stop-and-wait protocols proposed in [15] provide comparatively shorter data delivery delay and lower power consumption, sender still waits for acknowledgments or till time out period before making a retransmission attempt. Several protocols have been proposed in the literature that relies on pure FEC based schemes for improving the reliability of UASNs [16–19]. Even though, pure FEC-based schemes can reduce the end-to-end delay, such schemes compromise on effective data rate for achieving reliability.

To reduce the delay and to improve the reliability, hybrid ARQ schemes that exploit the advantages of both FEC and ARQ have been widely explored for UASNs [20–26]. In [20], the authors employ fountain code to enhance the reliability and efficiency of broadcasting in UASNs, while in [21], the authors use fountain codes for reliable data transport and storage. In [22], the authors propose fountain code-based adaptive multi-hop reliable data transfer (FOCAR), a hybrid ARQ scheme which integrates Fountain codes with hop-by-hop retransmission upon failure, for reliable data transfer in UASN. In [23], the authors propose a practical coding multi-hop reliable data transfer (PCMRDT) protocol for UASN that combines random linear coding and hop by hop selective repeat retransmission scheme to achieve high reliability. In [24], the authors provide experimental results for a hybrid ARQ scheme known as UW-HARQ, which combines random binary linear coding based FEC and ARQ. In [25], the authors propose segmented data-reliable transport (SDRT) protocol to achieve reliable data transfer in UASNs by employing Tornado code as the FEC scheme. In SDRT, after the transmission of an encoded block of data packets, the sender wait for an ACK and if it does not obtain an ACK from the receiver, it will send one more encoded packet to the receiver after the time out period. This procedure is continued until an ACK is received. However, in SDRT, during the retransmission phase, sender does not have the information about how many more packets are required for successful decoding at the destination node. Further, tornado code requires transmission of more redundant blocks for satisfying a specified reliability criterion, leading to higher end-to-end delay. In [26], the authors model the underwater channel as a Finite State Markov Chain (FSMC) and evaluate the performance of hybrid incremental redundancy ARQ protocols for a two node scenario. In [27], the authors consider reliable data collection in single hop UASN while the focus on our paper is on reliable data transfer in multihop UASN. In [28], the authors design a cooperative scheme for UASN combining Orthogonal Frequency Division Multiplexing (OFDM) with dynamic coded cooperation (DCC). Two OFDM-DCC examples are presented, one based on non-binary low density parity check (LDPC) codes applied across OFDM blocks, and other using inter block erasure correction codes. However, the work in [28], employs erasure codes with fixed rate while we employ rate adaptive erasure coding in which the number of encoded packets required over each hop depends on the perceived packet error rate (PER) over the hop. Further, the study in [28] is for a two-hop network while we consider a multihop network.

Erasure codes have been extensively applied as pure FEC codes for improving the data transmission reliability in terrestrial wireless sensor networks [7, 8, 29, 30]. From our detailed literature survey, it has been concluded that no prior work has appeared on combining RS erasure code based FEC with selective ARQ in a multihop UASN. The aim of this paper is to design a hybrid ARQ scheme that employs hop by hop erasure coding technique to achieve reliability over each hop in a UASN. In this case, each sender node along the multi-hop path adaptively computes the number of redundant packets required to meet a given successful decoding probability criterion over its immediate neighbor hop based on the perceived packet error rate (PER) over the hop. The FEC scheme is integrated with hop by hop ARQ in which the sender-receiver pair of each hop exchange ACK and NACK packets. While the ACK packet indicates successful block transmission, the NACK packet from the receiver indicates decoding failure and can be used by the sender to find the number of additional packets required for successful decoding. The NACK packets can also be used to extract the current PER information of the corresponding hop. In this way, the source node can make correct decision on the number of packets to be sent out during the retransmission phase to guarantee that the next hop node is able to recover the message block.

3 Reed-Solomon packet level erasure codes

Erasure codes can be categorized as optimal and near-optimal erasure codes [31]. RS codes [32] are classified as optimal erasure codes, in which any subset of m packets (from the transmitted set of $m+k$ packets) are adequate to successfully decode the original data block of m packets at the receiver. With RS coding, the number of redundant packets (i.e., k) required to meet a given criterion on successful decoding probability at the receiver is directly related to the PER along the path from the transmitter to receiver. On the other hand, the near optimal erasure codes such as Tornado codes [33], introduce a slightly higher overhead such that $m' = (1+\varepsilon)m$ packets are required to decode the data successfully at the receiver, where $\varepsilon > 0$. Rateless erasure codes or fountain codes such as Luby Transform (LT) codes [34] and Raptor codes [35] can produce potentially very large number of encoded packets from a given set of message packets; however, they also require additional redundant packets for successful decoding. Accordingly, we do not consider near optimal and rateless erasure codes.

RS encoding procedure involves generating $(m+k)$ equations with m unknown variables. The decoding involves finding the m unknown variables through solving any m equations out of the $(m+k)$ original equations. The generated code vector (C) after RS encoding can be represented as $C = VD$, where V is the Vandermonde matrix and D represents message data vector. An $(m+k) \times m$ Vandermonde matrix is given in Eq. (1) with elements of the form $v(i,j) = \phi_i^{j-1}$, $\phi_i \; \varepsilon \; GF(q^r)$.

$$V = \begin{bmatrix} 1 & \phi_1 & \cdots & \phi_1^{m-1} \\ 1 & \phi_2 & \cdots & \phi_2^{m-1} \\ 1 & \phi_3 & \cdots & \phi_3^{m-1} \\ \vdots & \vdots & \ddots & \vdots \\ 1 & \phi_{m+k-1} & \cdots & \phi_{m+k-1}^{m-1} \\ 1 & \phi_{m+k} & \cdots & \phi_{m+k}^{m-1} \end{bmatrix} \tag{1}$$

Assuming the given data is divided into m messages, $d_0, d_1, \ldots, d_{m-1}$, the code word polynomial $C(\Phi)$ can be written as,

$$C(\Phi) = \sum_{i=0}^{m-1} d_i \phi^i \tag{2}$$

The code word polynomial $C(\Phi)$ should be assessed at $(m+k)$ different points : $\phi_1, \ldots, \phi_{m+k}$; which is represented as $c_0, c_1, \ldots, c_{m+k-1}$ in Eq. (3).

$$\begin{bmatrix} 1 & \phi_1 & \cdots & \phi_1^{m-1} \\ 1 & \phi_2 & \cdots & \phi_2^{m-1} \\ 1 & \phi_3 & \cdots & \phi_3^{m-1} \\ \vdots & \vdots & \ddots & \vdots \\ 1 & \phi_{m+k-1} & \cdots & \phi_{m+k-1}^{m-1} \\ 1 & \phi_{m+k} & \cdots & \phi_{m+k}^{m-1} \end{bmatrix} \begin{bmatrix} d_0 \\ d_1 \\ \vdots \\ d_{m-1} \end{bmatrix} = \begin{bmatrix} c_0 \\ c_1 \\ \vdots \\ c_{m-1} \\ c_m \\ \vdots \\ c_{m+k-1} \end{bmatrix} \tag{3}$$

At the receiver, decoding involves finding the data vector D such that $VD = R$, where R represents the received code vector. If we can collect any of the m code words at the receiver, then the corresponding m rows of the Vandermonde matrix can be used to obtain the original data vector D. To reduce the encoding and decoding complexity, we make use of some of the modifications suggested in [36] as described below.

3.1 Extension field operations

Extension fields or $GF(q^r)$ are finite fields with $p = q^r$ elements, with q prime and $r > 1$. By keeping $q = 2$, the operations in extension fields can be made more simple by transforming the addition and subtraction into bit-by-bit modulo 2 operations.

3.2 RS code in systematic form

Systematic encoding which is represented in Eq. (4), enables efficient information processing by reducing the computational complexity at the decoder and encoder side. At the encoding side, the use of systematic codes will reduce the overhead since no computations are required for the portion of code words containing original messages. At the decoder side, any subset of m code words is enough for successful data recovery. If all the received codewords directly correspond to the original message, the message can be recovered without any computation. Further, if most of the received code words contain original messages, the decoding matrix will become closer to an identity matrix, making decoding very fast.

$$\begin{bmatrix} 1 & 0 & \ldots 0 \\ 0 & 1 & \ldots 0 \\ \vdots & \vdots & \vdots \\ 0 & 0 & \ldots 1 \\ 1 & \phi_{m+1} & \ldots \phi_{m+1}^{m-1} \\ \vdots & \vdots & \vdots \\ 1 & \phi_{m+k-1} & \ldots \phi_{m+k-1}^{m-1} \\ 1 & \phi_{m+k} & \ldots \phi_{m+k}^{m-1} \end{bmatrix} \begin{bmatrix} d_0 \\ d_1 \\ \vdots \\ d_{m-1} \end{bmatrix} = \begin{bmatrix} d_0 \\ d_1 \\ \vdots \\ d_{m-1} \\ c_m \\ \vdots \\ c_{m+k-1} \end{bmatrix} \tag{4}$$

3.3 Multiple independent code words in a single packet

As described in [36], we adopt certain modifications for the way in which encoded packets are created for transmission. Here, each encoded packet contains multiple independent code words instead of a single long code word. If one full data packet is designed to carry one long code word, each code word length has to be selected to be very large (so as to match the size of the packet). This will increase the implementation complexity as far as RS encoding and decoding procedures are considered, since long code word generation requires operations on larger fields that demands very large memory space and time. At the same time, we cannot use small message packets and small code words since this would lead to very small payload size within a packet. This may not suit the application requirements and may affect the network performance considerably. By putting multiple independent code words into a packet, we can fully utilize payload space of a packet without having problems associated with generation of larger code word lengths.

To elaborate further, imagine dividing one big data into B small pieces of data chunks as shown in Fig. 3. Then each data chunk is again divided into m messages, and encoded into $m+k$ code words. As shown in Fig. 3, a given encoding unit consists of m messages getting encoded into $m+k$ code words. This is done by multiplying the $(m+k) \times m$ systematic Vandermonde matrix with $(m \times 1)$ message vector. The $m+k$ code words thus generated correspond to multiplication of $(m+k)$ rows of the Vandermonde matrix with the message vector and let these code words be assigned code word ID that corresponds to the $m+k$ rows of the Vandermonde matrix. Since the RS encoding employs arithmetic in the extension field $GF(2^r)$ as given by (4), we need r bits to represent the code word ID. Once the encoding process is over, we have total of $B(m+k)$ code words to send. A packet is formed by packing the code words with same code word ID into a single packet, i.e., ith code words from each independent data chunk are packed together to form the ith encoded packet. The encoded packets should carry the code word ID and we need to reserve r bits in the packet header for this. From the code word IDs of the received packets, the corresponding rows of the Vandermonde matrix can be regenerated at the receiver and decoding can be performed.

Algorithm 1 The Encoding Process

for *all* $p(0 \le p \le B-1)$, **do**
  take the $d_{i,p}$ *encoding unit*, $i = 0, 1, ..m-1$
  multiply with Vandermonde matrix of order $(m+k) \times m$
  such that
  $c_{i,p} = \sum_{j=0}^{m-1} v_{ij} d_{jp}, 0 \le i \le m+k-1$
end for
for *all* i, **do**
  Group all $c_{i,l}$ *to same packet*, $0 \le l \le B-1$
end for

3.4 Look up table construction

Look up tables are used in our work to avoid repeated multiplication and division operations. Addition is XOR of two numbers while for multiplication and division operations, exponent and log values are computed and stored as tables [36]. Here, encoding and decoding process do not involve complex operations.

4 Protocol description

In this section, we present the details of proposed erasure codes based hybrid ARQ scheme for UASNs. We consider a multihop path of UASN in which each sensor node is equipped with only one half duplex acoustic modem. We assume the network to be stable with very low mobility. When a sensor node has data to report, it transfers the data towards the sink node through multihop acoustic links.

4.1 Protocol overview

In the proposed hybrid ARQ scheme, encoded packets are transferred block-by-block and hop-by-hop. The scheme combines RS code based hop-by-hop erasure coding along with hop-by-hop selective retransmissions. The hop-by-hop erasure coding is employed to achieve reliability over each hop along the path. In this case, the source node and all the intermediate nodes along the multi-hop path perform the RS encoding process and the decoding is performed by all the nodes, except the source node. Assume that the block size (i.e., the number of original packets in each block) be the same for all nodes and let it be equal to m. To achieve the required level of reliability over hop i, node N_{i-1} adaptively decides the number of redundant packets to be transmitted (i.e., code rate) to its immediate successor node N_i based on the knowledge of PER along hop i and the successful decoding probability criterion. When node N_{i-1} receives packets from its predecessor node, it decodes and arranges the packets into blocks of size m and apply RS based erasure coding to generate m_i number of encoded packets; sends the coded packets to node N_i and switches to receiving status. Recall that, when erasure coding is performed on a block of m packets, any subset of m encoded packets are just enough to reconstruct the original data at the receiver, i.e., if node N_i gets any of the m encoded packets, it can correctly decode the original m data packets, after which it sends an ACK packet to node N_{i-1}. Otherwise, node N_i will send an NACK packet to node N_{i-1} that contains information regarding the number of packets received correctly. The NACK packets are further utilized to inform the sender how many and which data packets (i.e., ID of the missing packets) are missing at the receiver. The sender node N_{i-1} will then retransmit the required number of additional encoded packets with specified IDs to the receiver node N_i. This procedure continues till node N_{i-1} receives an ACK from N_i. The number of retransmissions across each hop can be reduced by ensuring that transmission of each block of data packets can be successfully decoded with high probability. Let δ be the unsuccessful decoding probability at node N_i and let $p_{e,i}$ be the PER corresponding to hop i. While choosing m_i, we need to ensure the following:

$$\sum_{k=0}^{m-1} \binom{m_i}{k} p_{e,i}^{m_i-k} \left(1-p_{e,i}\right)^k \leqslant \delta \tag{5}$$

Figures 1 and 2, respectively, depict the procedure to be followed by the source node N_0 and an intermediate node N_i. First of all, the source node N_0 groups m data packets into one block and encodes them into m_1 packets using systematic erasure coding. Here, we assume that the source node N_0 has already acquired the initial PER information corresponding to its link to Node N_1 (i.e., hop 1). To acquire the initial PER over hop 1, we assume that node N_0 send HELLO packets while node N_1 computes the bit error rate (BER) over hop 1 based on the received SNR information and by using an appropriate channel model for underwater acoustic link as reported in [37]. Node N_1 transfers the BER value through a HELLO response packet and N_0 computes the PER for a data packet of size Y bits as $PER = 1 - (1 - BER)^Y$. Even though this approach is valid for independent bit error case, it provides a means to get an approximate estimate of the initial PER so that encoding ratio can be fixed appropriately. The protocol assumes that all the nodes (except the sink node) collect the initial PER over the hops to their immediate successor nodes in a similar way. Upon receiving a packet in a block of transmitted data, node N_1 performs cyclic redundancy check (CRC) and if the CRC fails, the packet is considered as erroneous. Therefore, N_1 is capable of counting how many packets are received correctly in a given block of data packets. If node N_1 fails to receive m data packets correctly, it sends back NACK to the source node N_0 by embedding information regarding the number of packets correctly received in the last transmission of the block of encoded packets. Upon receiving the NACK, the source node can calculate PER as the ratio between the number of packets correctly received to the total number of encoded packets sent out in the last transmission. The NACK packets are further utilized to inform the source how many and which data packets are missing at the destination (i.e., ID of the missing packets) and therefore the source node can make correct decision on which encoded packets are to be retransmitted so that the receiver node N_1 is able to complete the decoding procedure. This procedure continues until an ACK packet is received at the source node N_0, indicating successful reception of all the m data packets at node N_1. This procedure is implemented at every intermediate node along the multihop path so that the entire message gets delivered reliably at the sink node.

Whenever an intermediate node receives NACK packet from its successor node, it will be able to update the PER information of the corresponding hop. This will help the node to fix the level of redundancy required for the transmission of next block of data. However, one major problem with this approach is that the number of lost packets in one transmission trial could be a random variable and hence current PER information cannot be considered for the accurate computation of the redundancy for the next block of data. To have a finer estimate of the PER, we have used the exponential weighted moving average (EWMA) method. Over hop i, we use the following equation to update the PER [38]:

$$\bar{p}_{e,i}(n) = \theta \bar{p}_{e,i}(n-1) + (1-\theta)\hat{p}_{e,i}(n) \tag{6}$$

where $\bar{p}_{e,i}(n)$ and $\hat{p}_{e,i}(n)$, respectively, are the estimated and inferred PER (i.e., sample PER) in the nth measurement time. The coefficient $\theta \epsilon (0, 1)$ is a constant that can be used to smoothen the PER estimate and through which

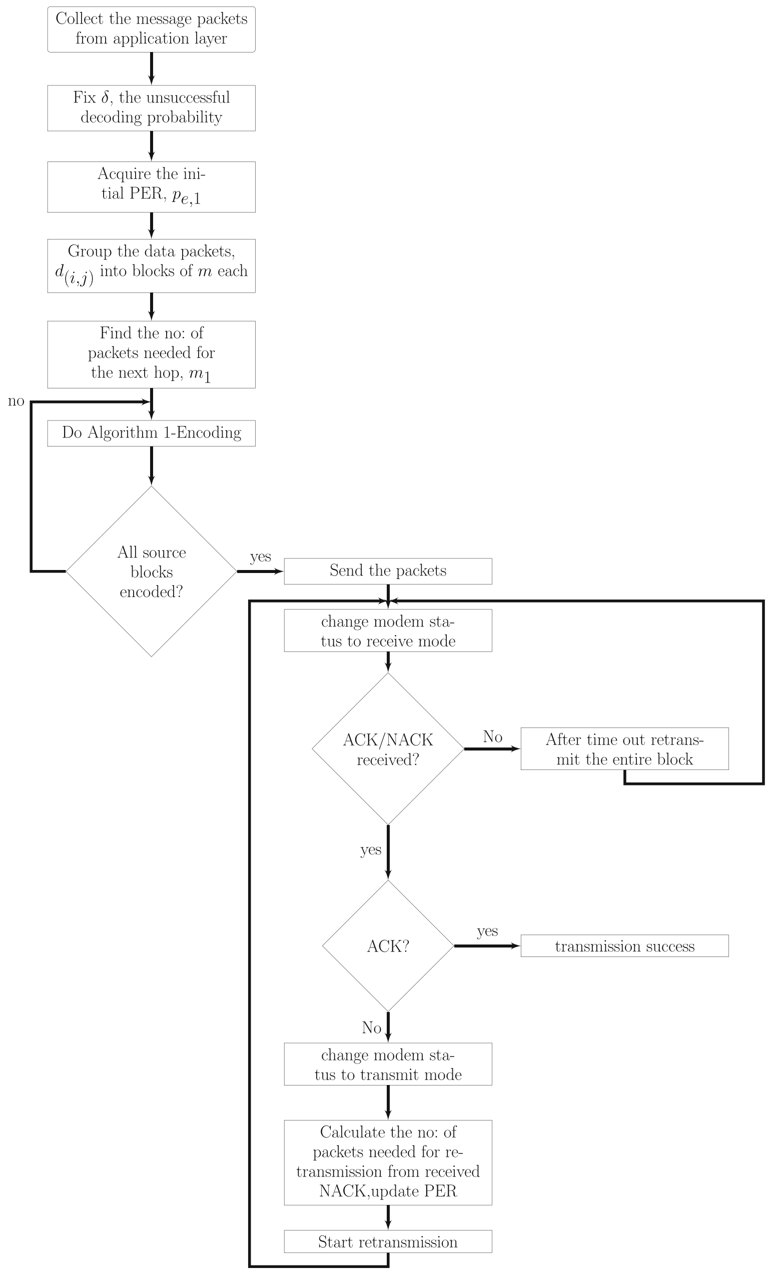

Fig. 1 Operations at the source node - N_0

we can control how quickly the influence of older transmissions decreases. It has been observed that a higher θ could be used for highly variable underwater acoustic channels, as it removes the effect of older transmissions faster. We have chosen $\theta = 0.9$ for the performance evaluation.

The above procedure provides a means to estimate the PER accurately and to calculate the number of redundant packets for the next transmission. If the PER varies highly dynamically, an estimation error may creep in. In this case, the receiver of the hop may fail to get sufficient number of packets to start the decoding process, triggering transmission of NACK packet from the receiver and selective retransmissions from the sender side. At the same time, the ACK/NACK packets may also be lost in the network. After the transmission of a block of encoded packets over a given hop, the sending node wait for a time duration equal to the round trip time (RTT) of that hop plus the transmission delay of ACK/NACK packets for receiving either an ACK or NACK. If the sender fails to get either

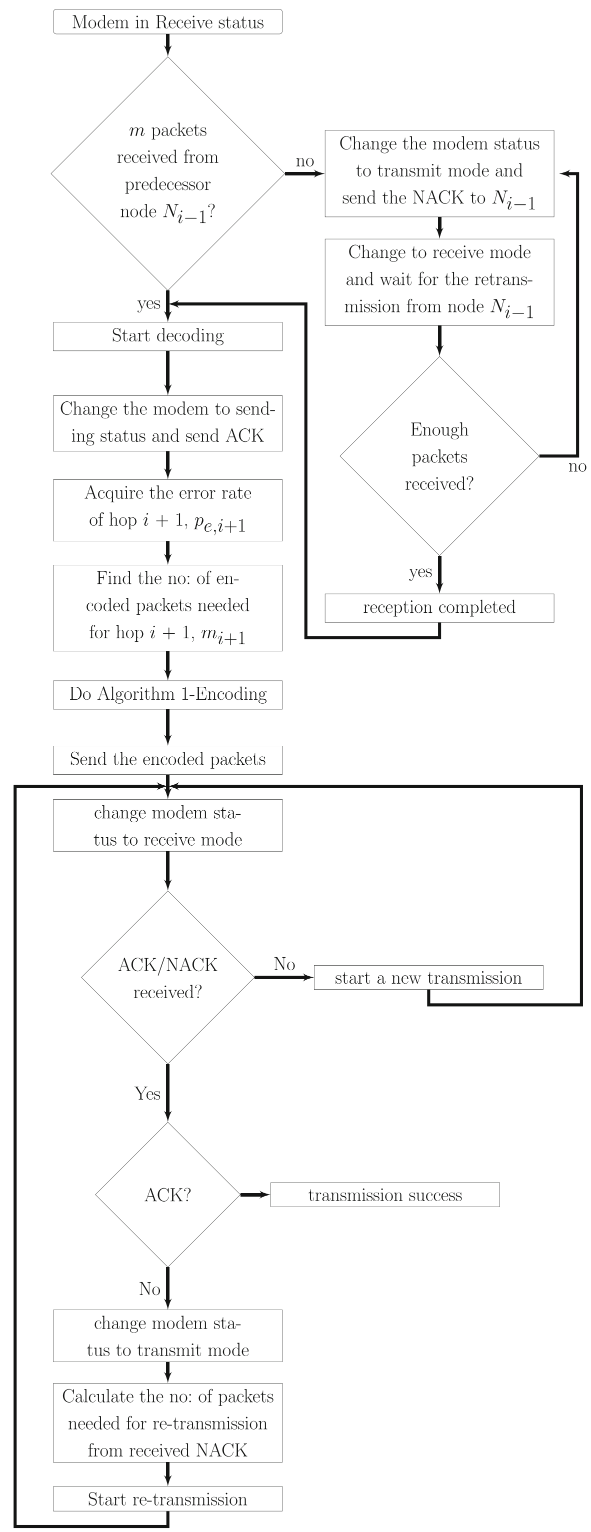

Fig. 2 Operations at the intermediate node - N_i

an ACK or NACK from the receiving node within the time out, it will retransmit the entire block of data. It is assumed that the sending node of every hop knows the distance to its receiver by using some appropriate range based technique. Updating the PER information with the help of NACK packets can reduce the total number of retransmissions in the network and hence the proposed scheme is more adaptable to the dynamic characteristics of the underwater acoustic channel. Since any subset of m packets is just enough to reconstruct the original message, node N_{i-1} can retransmit a subset of the original packets which have not yet received at the destination. As mentioned in Section 3, since systematic codes are used, when we have more received code words containing original messages as such, the decoding complexity would be reduced considerably.

In the design of the reliable data transfer protocol, we do not consider link layer FEC and assume that there is no interaction between the proposed hybrid ARQ scheme and the error detection and correction schemes employed at the lower layers. Generally, error detection is provided by the lower layers of the protocol stack (i.e., PHY and data link layers) which use checksums such as cyclic redundancy codes (CRC) to detect errors and discard corrupted packets. Further, error correcting codes are generally used at the PHY and data link layers (i.e., modems) to achieve a desired bit error rate performance in noisy links. Once error detection and correction processes are taken care of by lower layers, the upper layers of the protocol stack have to deal with packet erasures and losses that may arise due to many reasons such as network congestion, buffer overflow, etc. The uncorrectable errors at the data link layer due to severe multipath fading effects and interference can also be considered as equivalent to packet erasures. Since the aim of the current work is to improve the reliability of data transmission process in UASNs by effectively dealing with packet losses or erasures that can happen due to multiple reasons stated above, the proposed hybrid ARQ scheme needs to be implemented at the higher layer of the protocol stack (i.e., above the data link layer). Accordingly, in this paper, we do not consider error detection and correction which are generally implemented at lower layers of the protocol stack (Fig. 3).

5 Analysis

In this section, we provide models for finding the average delay and energy consumption of the network under the proposed scheme. For the analysis, we assume that, a multihop path exists from the source node (N_0) to the sink (N_L), which consist of $L+1$ nodes and L hops as shown in Fig. 4. Consider hop i, connecting nodes N_{i-1} and N_i. Given a message block consisting of m packets, sender N_{i-1} will transmit m_i encoded packets to node N_i and will enter the receive mode. Node N_i will try to decode the data and will enter the send mode and transmit either ACK or NACK depending on whether the decoding is successful or not.

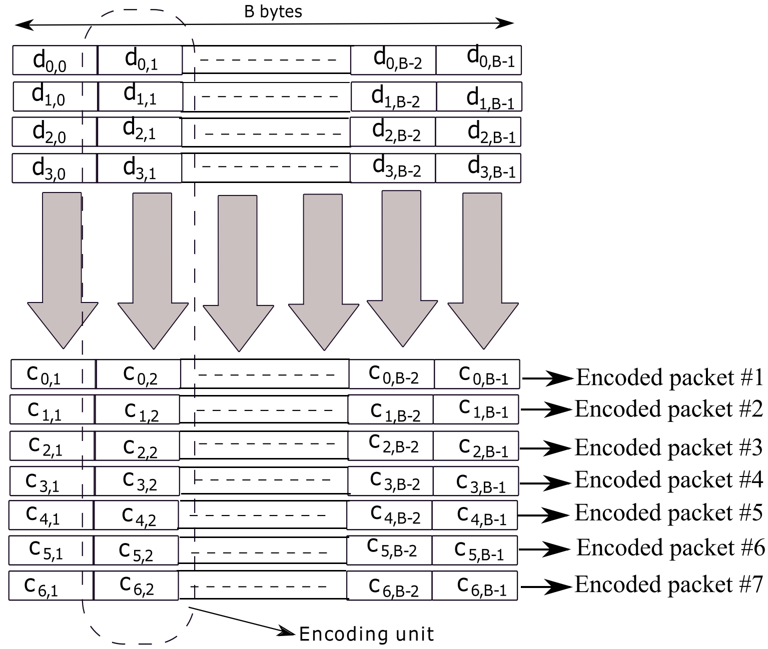

Fig. 3 Packet generation with multiple independent code words

5.1 Delay analysis

Consider hop i of the L - hop path from source to destination. The total delay experienced over hop i includes the encoding/decoding delays, transmission delay, propagation delay, and the modem transition delay. Assuming the number of transmission attempts till success to be geometrically distributed, the average number of transmissions required for the successful transmission of a message block is equal to $\frac{1}{1-\delta}$, where δ is the unsuccessful decoding probability over hop i. So, total delay incurred for the transmission of one message block over hop i is given by

$$T_i' \cong \frac{T_i}{(1-\delta)} + \frac{Y}{\alpha B(l_i)} m_i \tag{7}$$

The variable T_i is the sum of propagation time and transition time of the acoustic modem between the sending and receiving status (considering the encoding and decoding delay to be negligible compared to the propagation delay and the modem transition delays). Further, $\frac{Y}{\alpha B(l_i)} m_i$ is the average transmission delay, where Y represents the packet size, α is the bandwidth efficiency of the modulation and $B(l_i)$ is the bandwidth available for a link of distance l_i. From [39], we have

$$B(l_i) = b l_i^{-\beta} \tag{8}$$

where b is a constant expressed in dB relative to 1 kHz and β is an exponent expressed in dB per km.

The variable T_i is given by,

$$T_i = T_{prop} + 2T_{transition} \tag{9}$$

where T_{prop} is the propagation delay, which includes the round trip time for one transmission (includes the propagation delay for a block as well as for the ACK/NACK packet); $T_{transition}$ is the modem transition delay (for a single transmission, the modem needs to change the state two times). Thus, the total delay associated with the successful transmission of a block of m data packets over the L hop path is given as,

$$T_{total} = \sum_{i=1}^{L} T_i' \tag{10}$$

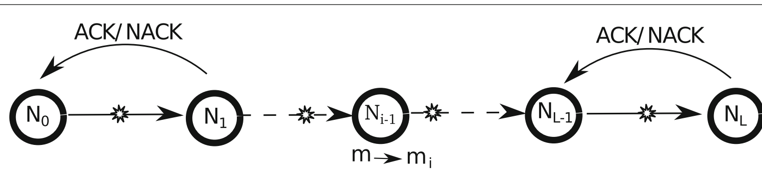

Fig. 4 Multihop path

5.2 Energy consumption analysis

In this section, we provide a model to determine the total energy consumption in sending a block consisting of m data packets over the network under the proposed hybrid ARQ scheme. Consider hop i along the L hop path, connecting source and destination nodes N_{i-1} and N_i, respectively. The various energy consumption components are (i) energy consumed for the data packet transmission and reception, (ii) energy consumed for ACK/NACK reception and transmission; and the total energy consumed is given by,

$$E_i = E_{tx,i} + E_{rx,i} + E^A_{rx,i} + E^A_{tx,i} \tag{11}$$

where $E_{tx,i}$ and $E_{rx,i}$, respectively, are the total energy consumption associated with the transmission and reception of the data block; $E^A_{tx,i}$ and $E^A_{rx,i}$, respectively, are the total energy consumption associated with the transmission and reception of the ACK/NACK packets. Now, $E_{tx,i}$ can be calculated as follows:

$$E_{tx,i} = m_i P_{tx,i} T^D_{tx,i} \tag{12}$$

where m_i is the average number of encoded packets required for successful delivery of the block of m packets along the hop i, $P_{tx,i}$ is the minimum transmit power required along hop i and $T^D_{tx,i}$ is the transmission time for a data packet, which is given as $T^D_{tx,i} = \frac{Y}{\alpha B(l_i)}$. The receive energy consumption for an encoded data packet having Y bits, $E^D_{rx,i}$, is about one-fifth of the transmit energy ($E^D_{tx,i}$), considering specifications of commercial hydrophones [40], and is given by

$$E^D_{rx,i} = \frac{1}{5} E^D_{tx,i} \tag{13}$$

After the reception of sufficient number of packets *i.e.*, m for successful decoding, the receiver will discard remaining packets in the same block. So, the total receive energy consumption over the hop i is given by

$$E_{rx,i} = m E^D_{rx,i} \tag{14}$$

The energy consumption due to ACK/NACK transmission, $E^A_{tx,i}$ can be given as,

$$E^A_{tx,i} = P_{tx,i} T^A_{tx,i} \frac{1}{1-\delta} \tag{15}$$

where $T^A_{tx,i}$ is the transmission time of ACK/NACK and $\frac{1}{1-\delta}$ is the average number of transmissions over hop i. Here, we assume that ACK/NACK packets are always delivered with very high reliability. The energy consumption due to ACK/NACK reception, $E^A_{rx,i}$ can be given as,

$$E^A_{rx,i} = \frac{1}{5} E^A_{tx,i} \tag{16}$$

Thus, the total energy consumption for an L hop system is given by,

$$E_{total} = \sum_{i=1}^{L} E_i \tag{17}$$

5.2.1 *Calculation of the minimum transmit power*

In this section, we describe an analytical model to compute the minimum transmit power required to satisfy a given target packet success transmission probability along hop i ($p_{s,i}$). We consider the L hop path from source to sink as shown in Fig. 4 and assume that the path has been formed after the route recovery phase. Instantaneous level of received power fluctuates as a result of small-scale fading effects due to multipath propagation in underwater environments. In shallow water, multipath occurs due to signal reflections from the surface, bottom and any objects in the water. In deep water, the multi path phenomenon occurs due to ray bending, i.e., the tendency of acoustic waves to travel along the axis of lowest sound speed. Even though, there is no general consensus on an appropriate statistical model for received signal amplitude in underwater acoustic channels, the small-scale fading effects are often modeled as Rayleigh or Rician fading [41–43] while some studies suggest K-distribution [44] or Weibull distribution [45]. In [46], Urick proposes Rician model to describe the amplitude fluctuations in ocean environment. In [47], Bjerrum -Niese et al. propose a turbulent shallow water channel model that follows Rician distribution. In this paper, we consider both Rayleigh and Rician fading models for the analysis of minimum transmit power. The analysis can be easily extended to other fading models as well.

Considering coherent BPSK as the modulation scheme, the bit error rate (BER) over Rayleigh fading channel is given by [48].

$$BER_i = \frac{1}{2}\left(1 - \sqrt{\frac{\gamma_{b,i}}{1+\gamma_{b,i}}}\right) \tag{18}$$

where $\gamma_{b,i}$ is the average SNR per bit. With BPSK over Rician fading channel, the average BER is given by [49],

$$BER_i = \frac{1}{2} erfc \sqrt{\frac{k\gamma_{b,i}}{k+\gamma_{b,i}}} \tag{19}$$

where k is the Rician factor and *erfc* is the complementary error function. The probability of successful transmission of a packet of size $'Y'$ bits over hop i is given by

$$p_{s,i} = (1 - BER_i)^Y \tag{20}$$

where BER_i is the bit error rate over hop i. For Rayleigh fading, an approximate equation for the minimum SNR per bit $\gamma_{b,i_{min}}$ required to satisfy a specified target for the probability of successful transmission per packet over hop

i (i.e., $p_{s,i}$) can be determined by combining (18) and (20) as follows:

$$\gamma_{b,i_{min}} \cong \frac{1}{4\left(1 - p_{s,i}^{\frac{1}{Y}}\right)} \tag{21}$$

Similarly, with Rician fading, the minimum SNR per bit $\gamma_{b,i_{min}}$ can be calculated by combining (19) and (20) as follows:

$$\gamma_{b,i_{min}} = \frac{k\left(erfc^{-1}\left(2\left(1-p_{s,i}^{\frac{1}{Y}}\right)\right)\right)^2}{k-\left(erfc^{-1}\left(2\left(1-p_{s,i}^{\frac{1}{Y}}\right)\right)\right)^2} \tag{22}$$

According to the passive sonar equation, the SNR per bit over an underwater acoustic link is given by [50, 51]

$$\gamma_{b,i} = SL_i - TL_i - NL(f) + DI \tag{23}$$

where SL_i is the source level that determines the transmit power over hop i; TL_i denotes the transmission loss corresponding to hop i; $NL(f)$ is the noise level; DI is the directivity index (all quantities assumed to be in dB). We consider omni directional hydrophone so that DI is zero. Now transmission loss (TL_i) over a distance $'l_i'$ (in meters) for underwater channel is given by [51]

$$TL_i(l_i,f) = \chi 10\log(l_i) + a(f)l_i \times 10^{-3} + A \tag{24}$$

where $a(f)$ is the absorption coefficient corresponding to a transmitting frequency f kHz; χ is the spreading coefficient (has value 1 and 2 for shallow and deep water scenarios, respectively); and A is the transmission anomaly [52]. Several empirical formulas exist for the absorption coefficient in literature; here, we adopt the Thorp's formula for simplicity and is given by [50–53]:

$$a(f) = 0.11\frac{f^2}{1+f^2} + 44\frac{f^2}{f^2+4100} + 2.75\times 10^{-4}f^2 + 0.003 \tag{25}$$

In the underwater case, the ambient noise is composed of four components: turbulence, shipping, waves, and thermal noise [39]. The empirical formulas for the power spectral densities (expressed in dB re 1μ Pa per Hz) of the four noise components are,

$$10\log N_t(f) = 17 - 30\log f \tag{26}$$

$$10\log N_s(f) = 40 + 20(s-0.5) + 26\log f - 60\log(f+.03) \tag{27}$$

$$10\log N_w(f) = 50 + 7.5\omega^{\frac{1}{2}} + 20\log f + 40\log(f+0.4) \tag{28}$$

$$10\log N_{th} = -15 + 20\log f \tag{29}$$

where $N_t(f)$, $N_s(f)$, $N_w(f)$ and $N_{th}(f)$ respectively denote the PSD's of turbulence noise, shipping noise, waves noise and thermal noise with f in kHz. The total noise PSD is the sum of different spectral densities as above. Different frequency ranges influence different Noise components. The overall PSD of the ambient noise level can be approximated as [39],

$$NL(f) \approx 50 - 18log_{10}f \tag{30}$$

From Eqs. (23), (24), and (30),

$$\gamma_{b,i_{min}} = SL_i - \left(\chi\, log(l_i) + a(f)l_i \times 10^{-3} + A\right) - (50 - 18log_{10}f) \tag{31}$$

The sound intensity of a source is related to a reference intensity and is given by [50, 51]

$$I_{tx,i} = 10^{\frac{SL_i}{10}} I_{ref} \tag{32}$$

where $I_{ref} = \frac{q^2}{2\rho c}$. Here, q is the effective sound pressure, ρ the density of seawater, and c is the propagation velocity of the sound wave in sea water. Assuming $c = 1500$ m/s, $\rho = 1000kg/m^3$, and $q = 1\mu Pa\ rms$, we have $I_{ref} = 0.67 \times 10^{-18}W/m^2$ [54].

In the case of deep water scenario, where spherical spreading is experienced, the power $P_{tx,i}$(watt) required to achieve sound intensity $I_{tx,i}$ at l_i meters from the source in the direction of the receiver is expressed as [51]

$$P_{tx,i} = 4\pi l_i^2 I_{tx,i} \tag{33}$$

For the shallow water scenario where cylindrical spreading occurs, the transmit power $P_{tx,i}$ is given by [51]:

$$P_{tx,i} = 2\pi l_i H I_{tx,i} \tag{34}$$

where H is the depth of the water in meters. Given a requirement for probability of successful transmission of a packet per link, the minimum SNR per bit required can be determined from Eqs. (21) and (22) for Rayleigh and Rician fading, respectively. Combining Eqs. (21) and (22) with Eq. (31), we can find the minimum SL_i required to satisfy the given constraint on $p_{s,i}$ for both deep and shallow waters. The minimum transmit power $P_{tx,i}$ required to satisfy the given constraint on $p_{s,i}$ over hop i can be determined using Eqs. (33) and (34) for deep and shallow water scenarios, respectively, by combining them with (32).

6 Results and discussion

In this section, we evaluate the performance of the proposed erasure codes-based hybrid ARQ scheme. We compare the performance of the proposed scheme against existing schemes such as selective ARQ, SDRT [25], and PCMRDT [23] in terms of the following parameters: (i) total number of packets that need to be sent in the network to ensure successful delivery of the given block of message packets; (ii) total energy consumed in the network; (iii) end-to-end delay experienced, and (iv) throughput. We have implemented the proposed scheme using Aquasim, an ns-2 based simulator for underwater acoustic networks [55]. We simulate a multi hop UASN consisting of five hops. The distance of each hop is uniformly distributed between 2 and 5 km. The transition delay between the sending and receiving status of the acoustic modem is selected to be 1.5 s, modem synchronization delay is 0.5 s and forwarding delay is 1 s [22, 23]. The initial energy of each node is assumed to be 70 J [56]. A four-state Markov model has been used to represent the acoustic channel over each link. Unless specified otherwise, the packet erasure rate of each acoustic link has been assumed to take any of the four values: 0.1, 0.2, 0.3, and 0.4. The link stays in each state for a time period that is exponentially distributed with mean value 200 milliseconds [57]. It is assumed that there is only one source node in the network with 200 original data packets, each of size 60 bytes. Further, four packets are grouped into a block and thus there are 50 original data blocks. We select the decoding failure probability (i.e., δ) to be 0.05 which means that, for transmission of each block of data, we try to ensure that the decoding success probability is almost 95%. We fix the size of HELLO/ACK/NACK packet to be 6 bytes each. The simulation duration has been fixed as 1000 seconds and the results are depicted by finding the average of 30 simulation runs. Other simulation parameters are given in the Table 1.

In selective ARQ, the sender transmits a window of packets and waits for the ACK corresponding to each packet. The sender then selectively retransmits those packets which are lost in the network. We consider the window size to be equal to 5 for the selective ARQ. In Figs. 5 and 6, we compare the proposed hybrid ARQ scheme, selective ARQ, SDRT, and PCMRDT in terms of total number of packets sent in the network for the given scenario. In Fig. 5, we plot this quantity as a function of hop count while the average erasure rate across each link between the source-destination pair is equal to 0.4. For all the schemes considered, the number of packets to be sent increases as the hop count is increased. However, the number of packets transferred is less in the proposed scheme as compared to all other schemes. Selective ARQ is a pure ARQ scheme and thus rely on packet retransmissions for reliable data delivery. The SDRT protocol sends a stream of encoded packets till it receives a positive feedback from the receiver and during the retransmission phase, sender does not have the information about how many more packets are required for successful decoding at the destination node. The PCMRDT is based on random linear coding; and it requires all the $(m + k)$ coded packets to be successfully received at the receiver to recover the original messages (i.e., m packets) which leads to more retransmissions. The proposed hybrid ARQ scheme employs dynamic computation of the code rate based on the perceived error rate across each link such that original data can be recovered with less number of retransmissions. Figure 6 gives the total number of packets sent as a function of average packet erasure probability. As the erasure rate increases, the number of packets needed for successful data delivery increases for all the schemes considered. However, in the proposed scheme, the total number of packets to be sent in the network is less as compared to all other three schemes. This is because, the proposed scheme computes the coding ratio adaptively and hence the average number of transmissions required per data block is significantly lower.

Table 1 System parameters for analysis and simulations

Parameter	Value
Sound speed profile (SSP)	Coppens equation [58]
Depth of shallow water	less than 100m
Temperature (shallow)	22 °C
Salinity (shallow)	35 parts per thousand
Depth of deep water	Between 500 and 1000 m
Temperature (deep)	15–12 °C
Salinity (deep)	35 parts per thousand
Constant b in Eq. (8)	14 dB
Constant β in Eq. (8)	0.5 dB/km
Operating frequency range	10 to 15 kHz
Bandwidth efficiency (α)	1 bps/Hz
Attenuation model	According to Eqs. (24) and (25)
Spreading coefficient	1 for shallow and 2 for deep water
Multipath fading type	Rayleigh or Rician

Figure 7 depicts the variation in the total number of packets sent against the packet length used. In this case, the source node is assumed to have 200 packets with packet length varied from 30 to 70 bytes. As the packet length increases, they become more susceptible to errors leading to more retransmissions. However, the proposed scheme has less number of retransmissions compared to that of other schemes considered and hence total number of packets transferred in the network is less.

Figures 8 and 9 show the total energy consumed in the network considering both Rayleigh and Rician fading for deep and shallow water scenarios respectively. We select the transmit power so as to achieve a packet success probability of 0.6. As the number of hops increase, the total energy consumed becomes higher, owing to larger number of re-transmissions in the network. The target SNR required in the case of Rayleigh will be higher than Rician fading because of which the total energy consumed will be higher for Rayleigh. Also the spherical spreading of the signal in deep water causes more power consumption and higher error rate than the shallow water scenario.

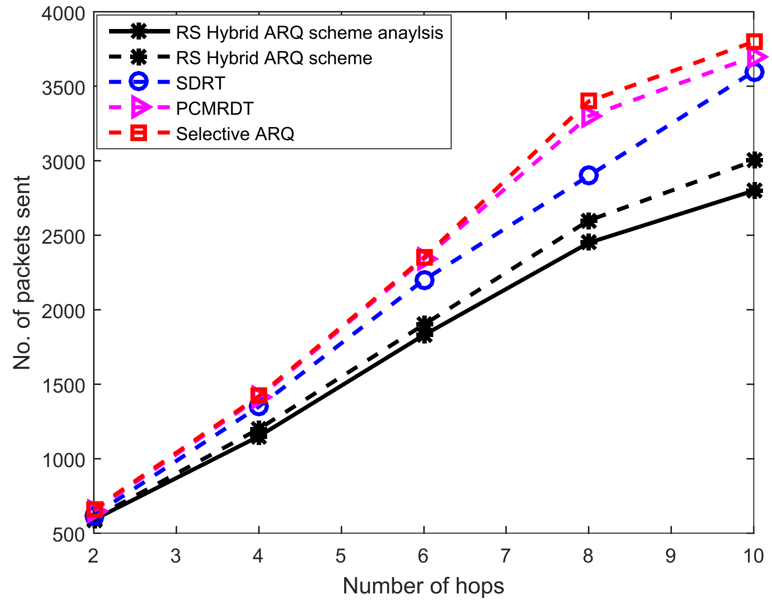

Fig. 5 Number of packets sent (average erasure rate = 0.4)

Accordingly, the energy consumption is higher for the deep water case. The total energy consumed is less for the proposed scheme compared to the other schemes under consideration, owing to the fact that, the average number of retransmissions per message block is less in the proposed scheme. Figures 10 and 11 depict the variation of the total energy consumed against the hop distance for Rayleigh and Rician fading, respectively, considering deep and shallow water scenarios. The available bandwidth for the underwater channel decreases as the distance between hops increases, because of which the transmission time increases, leading to an increase in the energy consumption. Further, as the distance increases, more power is needed for a successful transmission. Accordingly, the

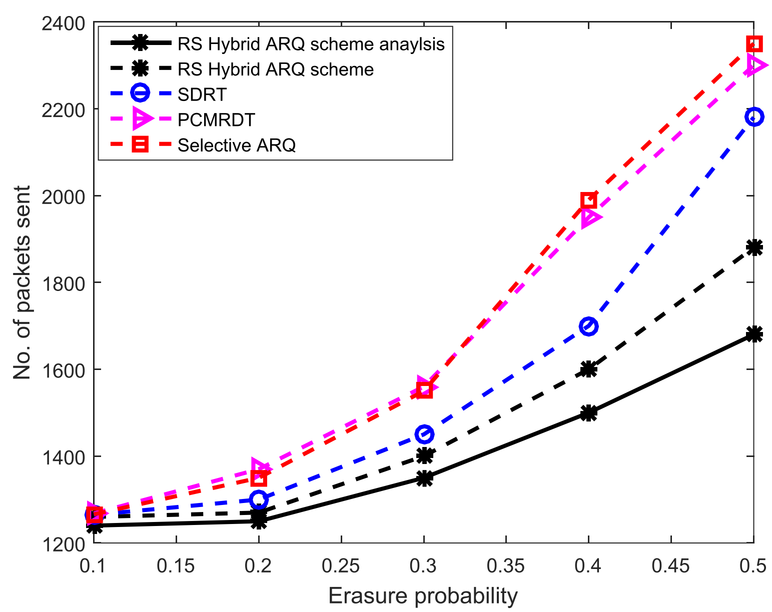

Fig. 6 Number of packets sent (number of hops = 5)

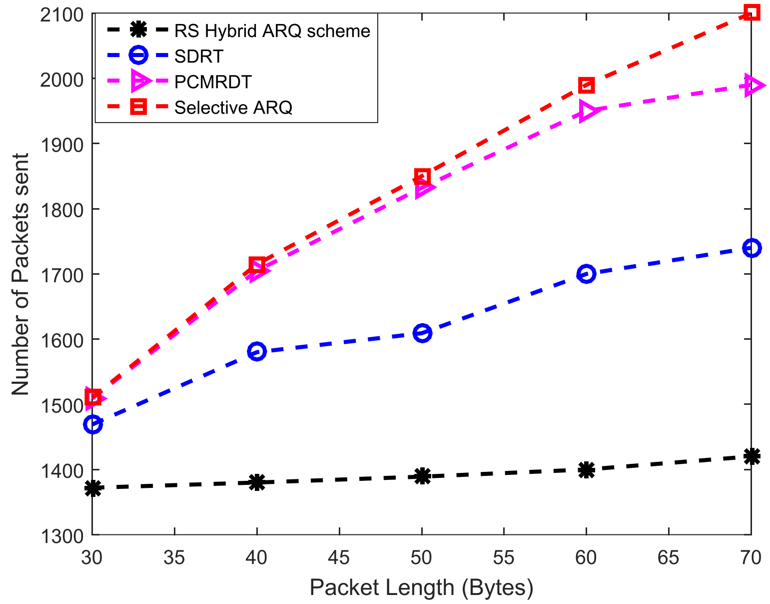

Fig. 7 Packets sent vs packet length

energy consumption graph shows an increasing trend as the hop distance increases.

Figures 12 and 13 depict the average end-to-end delay incurred for the transmission of one message block (consisting of four packets each of size 60 bytes) against the number of hops and the erasure probability, respectively. More number of retransmissions leads to the accumulation of propagation delays and modem transition delays and finally increase in the average end-to-end delay for selective ARQ scheme. In SDRT, the sender will retransmit one encoded packet, if it fails to receive ACK for a transmitted data block and the procedure is continued till it gets ACK. This results in very high end-to-end delay. For PCMRDT, all the $(m + k)$ encoded packets are

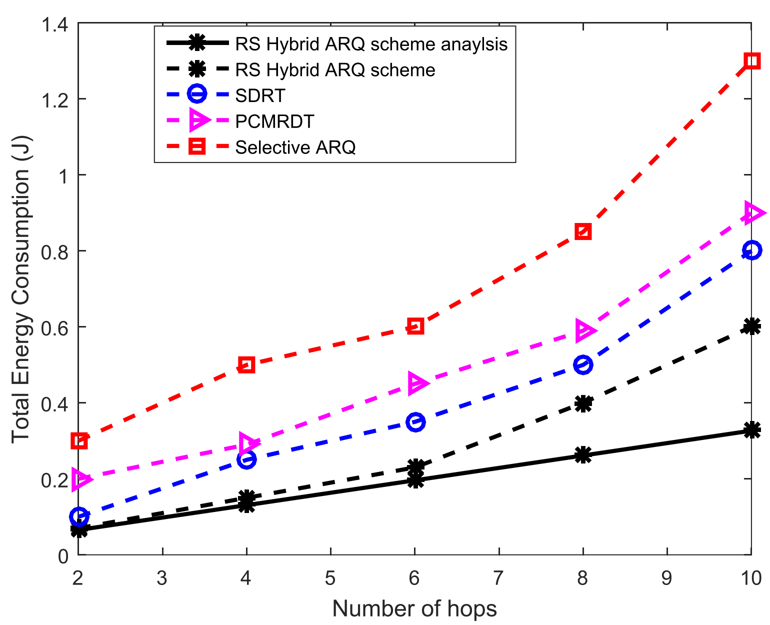

Fig. 8 Total energy consumption (Rayleigh fading-deep water)

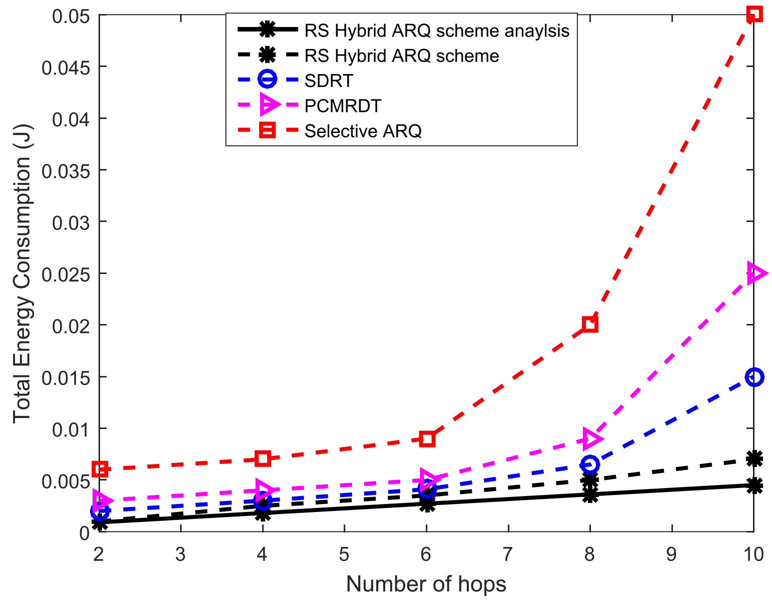

Fig. 9 Total energy consumption (Rician fading-shallow water)

needed at the receiver for successful recovery of the m message packets. In an error prone channel, this leads to larger number of retransmissions and higher end to end delay. With increase in erasure rate, the delay increases further due to increase of retransmissions. The proposed scheme tries to minimize the number of retransmissions by following an adaptive approach for computing the total number of encoded packets required per message block. Accordingly the end-to-end delay gets reduced significantly. Figure 14 shows the variation of the end to end delay against the hop distance. As the hop distance increases, the total available bandwidth for transmission in underwater acoustic channel decreases making the transmission time higher, leading to higher end to end

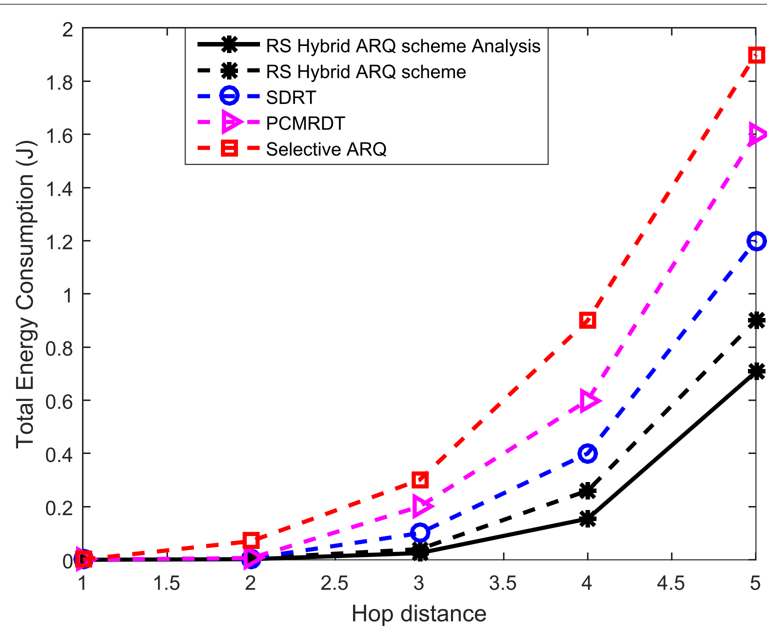

Fig. 10 Total energy consumption vs hop distance (Rayleigh fading-deep water, no. of hops = 2)

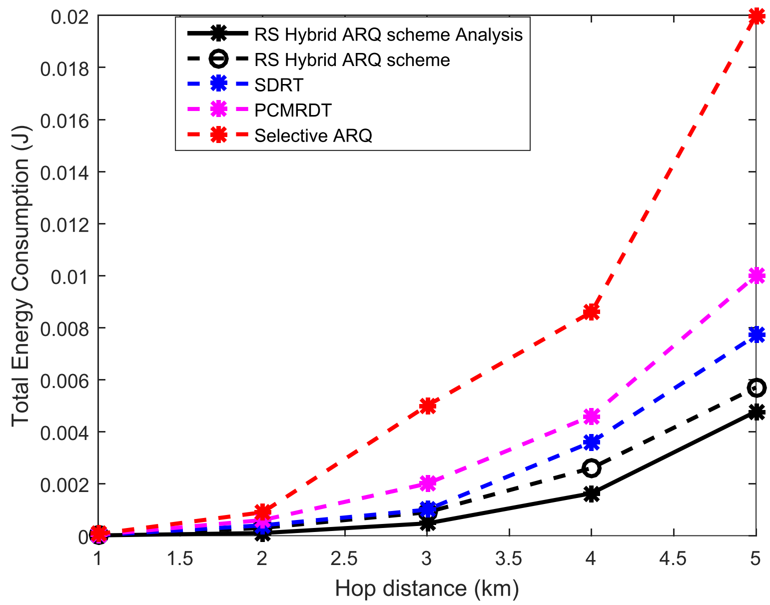

Fig. 11 Total energy consumption vs hop distance (Rician fading-shallow water, no. of hops = 2)

delay. However, the end to end delay has been observed to be less for the proposed scheme.

Figure 15 compares the throughput of the proposed scheme against that of other schemes considered in this paper. Throughput denotes the amount of successfully delivered bits per second at the sink node and is computed by finding the ratio of message block size delivered at the sink and the time taken for the delivery. The results show that the throughput decreases as the erasure rate becomes higher owing to the large number of re-transmissions required in the network and consequent increase of end-to-end delay. However, the throughput of the proposed scheme has been observed to be higher as compared to that of other schemes. This is due to the fact that the proposed scheme leads to less number of redundant transmissions and consequently the end-to-end delay incurred

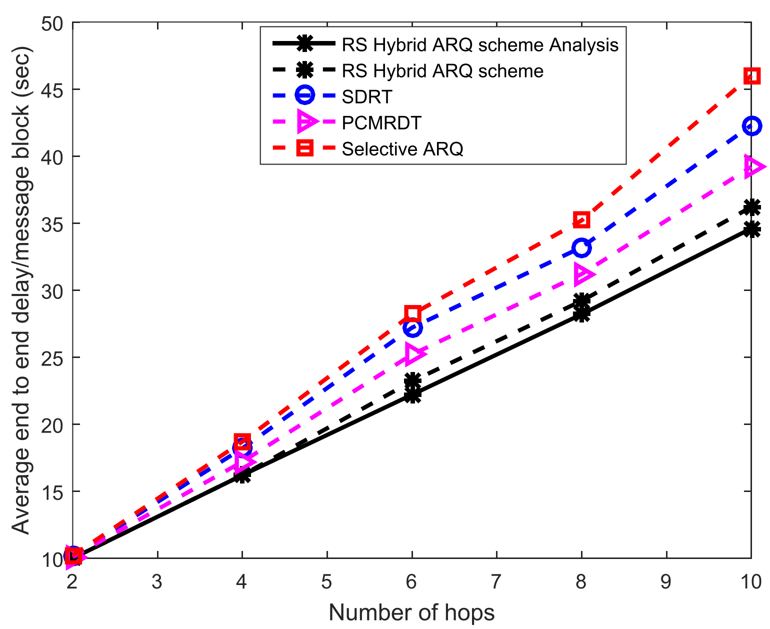

Fig. 12 Average end-to-end delay vs number of hops

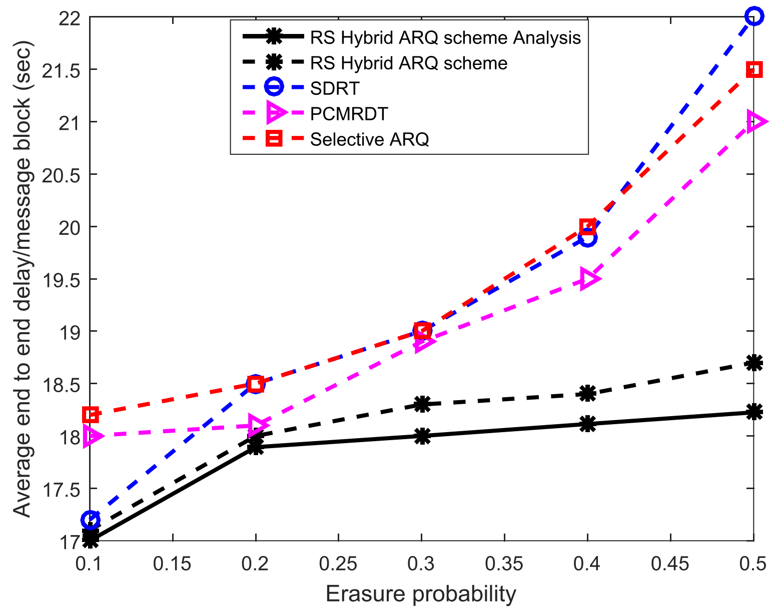

Fig. 13 Average end-to-end delay vs erasure probability (no. of hops = 5)

for the successful delivery of the message block is reduced. At the same time, adaptive computation of the code rate ensures higher packet delivery ratio as well.

7 Conclusions

In this paper, we have proposed a hybrid ARQ scheme that combines the Reed-Solomon-based erasure coding and selective retransmissions, for reliable data transfer in underwater acoustic sensor networks (UASNs). We have also described a method to compute the number of encoded packets required for each message block. The proposed scheme was implemented using Aquasim simulator and the simulation results have demonstrated that it performs significantly better than other existing

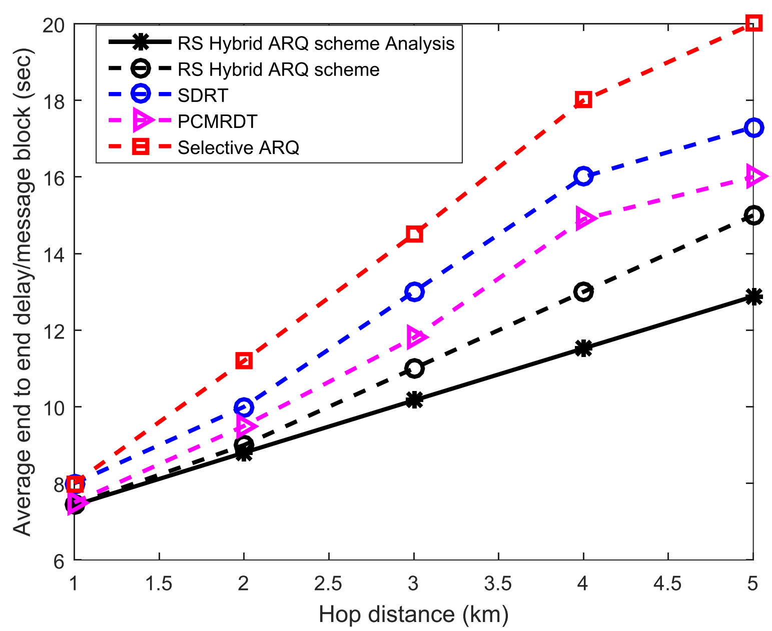

Fig. 14 Average end-to-end delay vs hop distance (no. of hops = 2)

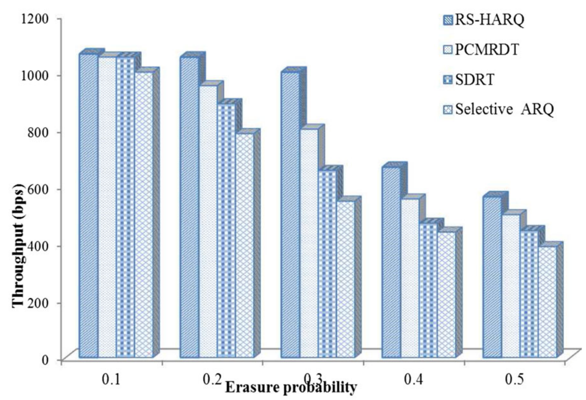

Fig. 15 Throughput (number of hops = 1, packet size =125 bytes, hop distance = 1 km)

schemes for reliable data transfer, in terms of metrics such as throughput, energy consumption and average delay.

Competing interests
The authors declare that they have no competing interests.

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