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ORIT: A Transport Layer Protocol Design for Underwater DTN Sensor Networks

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ABSTRACT Research challenges have focused on underwater acoustic sensor networks characterized by long propagation delay, narrow bandwidth, and frequently disconnected interruption, among others, which can be viewed as delay/disruption tolerant networks (DTNs). However, limited research has been conducted on goodput performance considering the total data packets transmitted in the channel, which is referred to as the remove redundant goodput (RRG) performance of underwater DTN sensor networks. In this paper, we propose a transport layer protocol, called optimal retransmission timeout (RTO) interval stop-and-wait transmission (ORIT), used in underwater DTN sensor networks. In most general transport layer protocols, the RTO timer is set to longer than the RTT to avoid pseudo-retransmissions. However, great network latency could be introduced by applying such a mechanism in underwater communication networks. In ORIT, we propose an RTO optimization algorithm to maximize the RRG performance of the network by reducing the RTO timer. Meanwhile, we adopt the interval stop-and-wait transmission mechanism to avoid data pseudo-retransmissions caused by setting the RTO timer shorter than the RTT in narrow acoustic channels. Through adopting the RTO timer, the ORIT can achieve the best RRG performance under an RTO timer, which we call the optimal RTO timer. We compare the ORIT with other transport layer protocols in the DTN network. The results show that the ORIT with the optimal RTO timer can shorten the long propagation delay and effectively increase the data delivery rate in the underwater DTN sensor networks.

INDEX TERMS Underwater acoustic sensor networks (UASNs), delay/disruption tolerant networks (DTNs), optimal retransmission timeout (*RTO*), remove redundant goodput (RRG).

I. INTRODUCTION

A. BACKGROUND AND RESEARCH CHALLENGES

Underwater acoustic sensor networks (UASNs) are underwater surveillance network systems consisting of a large number of sensor nodes with acoustic communication and computing abilities. In recent years, underwater acoustic communication technology has become a popular research topic that is widely used in a variety of applications such as ocean data collection, pollution control, offshore exploration, disaster prevention, assisted navigation and tactical observation [1]–[4]. Different from terrestrial sensor networks, due to the rapid attenuation of radio waves and the scattering effect of light waves, UASNs communicate mainly through acoustic signals. Since underwater acoustic channels have the characteristics of long propagation delay, narrow bandwidth, high noise, high bit error rate and propagation loss, UASNs can be viewed as delay/disruption tolerant networks (DTNs) [5].

The DTN was proposed as a network architecture that enables effective communication in a restricted network environment. It provides transmission guarantee for users to communicate in a heterogeneous network with the features of long delay, high error rate, frequently disconnected interruption, etc [6]. Thus, the DTN protocol can be applied in many restricted communication scenes such as interstellar networks, vehicle networks and underwater sensor networks [7]. In UASNs, the propagation speed of acoustic signals in seawater is approximately 1500 m/s, which is five orders of magnitude lower than the speed of radio propagation speed (3×10^8 m/s) [8]. Therefore, the particularly low propagation speed has led to extremely long data delivery delay, increasing the round-trip time (*RTT*) [9] (defined in TCP) in

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underwater DTN sensor networks. In the reliable data transmission protocol, an important mechanism is retransmission timeout (*RTO*). The sender starts a timer when data is sent. If the sender does not receive the acknowledgment (ACK) message when the *RTO* timer expires, it will retransmit the data packet until the data are delivered successfully. In most protocols for terrestrial communication, the *RTO* timer is set to longer than the average *RTT* to ensure that the retransmission occurs after the completion of the last data round-trip transmission to avoid unnecessary retransmission. When the sender receives ACK before the expiration of the *RTO* timer, it repeals the *RTO* timer that is already set. Otherwise, the sender considers the data lost and retransmits. If the *RTO* timer is set too long, there will be a long wait time before retransmission of the possibly missing data, which leads to low communication efficiency. However, if the *RTO* timer is set too short, there will be unnecessary retransmissions. Therefore, the appropriate setting of the *RTO* timer is important to balance the long wait time and unnecessary retransmissions. As mentioned above, because of the long *RTT* in underwater acoustic communication, setting the *RTO* timer according to the traditional terrestrial protocols will lead to low communication efficiency. Although numerous works have focused on the setting of the *RTO* timer in terrestrial communication networks, for example, the adaptive optimized *RTO* algorithm in [10] to improve the data transmission rate as well as an approach that changes the *RTO* calculation mechanism of SCTP in [11] to improve the throughput performance of wireless communication, these schemes only adjust the *RTO* timer with the dynamic change of the *RTT* and do not solve the long *RTT* problem to effectively improve the data transmission rate in underwater DTN sensor networks.

In this paper, we propose a transport layer protocol, called Optimal *RTO* Interval Stop-and-Wait Transmission (ORIT) and explore the *RTO* timer setting for the use of underwater acoustic communication in extremely long propagation delay and narrow channel environments, which can effectively increase the network data delivery performance.

B. ORIT OVERVIEW AND CONTRIBUTIONS

The main idea of ORIT is to maximize the goodput performance of underwater DTN sensor networks with an optimal *RTO* timer while transmitting as few redundant data packets as possible. Therefore, the total data packets transmitted in the channel should be considered. To evaluate the network performance, an evaluation of goodput, which is defined as the remove redundant goodput (RRG), is applied. Differing from most transport layer protocols used in terrestrial networks, in which the *RTO* timer is set to longer than the *RTT*, the optimal *RTO* timer in underwater acoustic communication has been reduced. At the same time, ORIT adopted the interval stop-and-wait transmission mechanism to solve the problem of data pseudo-retransmissions caused by the setting of the optimal *RTO* timer being shorter than the *RTT*.

As a whole, ORIT is adopted as the transport layer protocol in DTN structure to improve network performance.

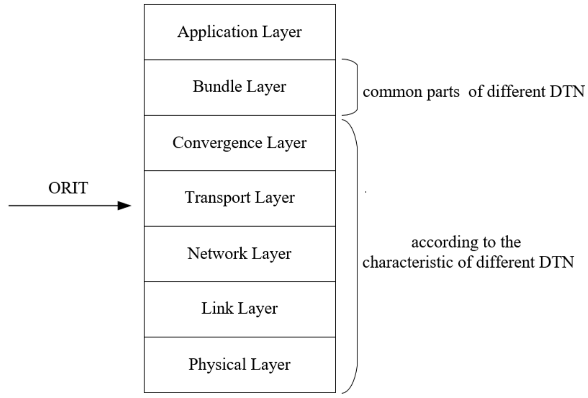

FIGURE 1. DTN protocol structure.

In underwater DTN sensor networks with long propagation delay, high error rate and frequently disconnected interruption characteristics, the DTN protocol can provide reliable data transmission guarantee through the store-and-forward mechanism and custody transfer mechanism. Owing to the long propagation delay, we propose ORIT as the transport layer protocol in DTN structure to shorten the network latency with the expectation that it can result in the network achieving the best RRG performance. In this protocol mechanism, ORIT and bundle layer protocol are mutually reinforcing that bundle layer can provide reliable data transmission guarantee and ORIT can shorten the long propagation delay for UASNs. Fig. 1 shows the DTN protocol structure and the location of ORIT in DTN. For different convergence layer protocols, transport layer protocols including TCP and UDP are run below them [12]. The convergence layer protocol UDPCL is an unreliable transmission protocol and has no confirmation mechanism. Thus, the RDT protocol is always used as a link layer transport protocol to ensure reliable communication for the UDPCL protocol [13].

The following is the main contribution to the subject. A transport layer scheme, ORIT, was proposed in the DTN protocol structure for underwater acoustic communication networks to optimize network performance. Based on the interval stop-and-wait transmission mechanism, an optimal *RTO* timer shorter than the *RTT* was found to achieve the best RRG performance for the communication network.

The remainder of this paper is organized as follows. First, we analyze the existing transport layer protocols in underwater communication networks and introduce ORIT in the DTN structure in section II. Then, the description of the ORIT execution rule is provided and the theoretical model of ORIT is built up in section III. Next, the RRG performance results of ORIT and other transport layer protocols in DTN with respect to the setting of the *RTO* timer are evaluated in section IV. Finally, the summary and conclusion of the paper are presented in section V.

II. RELATED WORK

In recent years, plenty of transport layer protocols have been proposed for DTN networks. In terrestrial networks, Fabio Albini *et al.* in [14] introduce a novel mechanism which is based on the delay tolerant transport protocol (DTTP) to improve ontent distribution in DTNs. The main idea of the mechanism is controlling the diversity in the coded information transmission without increasing the use of network resources or including any feedback messages. In interstellar networks, Ruhai Wang *et al.* in [15] propose an optimal RTO timer technique with RTO timer shorter than RTT to improve transmission efficiency of DTN protocol. Although both of interstellar networks and UASNs have the characteristic of long propagation delay, the premise of the mechanism for interstellar networks is assuming that the channel bandwidth is effectively sufficient, which is not suitable for UASNs with narrow bandwidth.

In UASNs, several transport layer protocols have been proposed to improve network performance. In the stop and wait ARQ protocol [16], acknowledgement is required to guarantee reliable communication before retransmission of the lost data packets, which can result in extra propagation delay. The improved ARQ scheme [17] still has to wait for acknowledgements even if with reduced delivery delay. To improve transmission performance, two main types of transport layer protocols are proposed, one is based on end-to-end encoding and the other is based on adaptive congestion window (cwnd) [1].

Following are the end-to-end encoding-based protocols. To improve data transmission reliability and reduce the long propagation delay, Peng Xie *et al.* in [18] propose SDRT, which is a hybrid approach combining the ARQ and FEC protocols. SDRT adopts efficient erasure codes as well as the window control mechanism, transferring encoded packets block-by-block and hop-by-hop. SDRT significantly reduces the total number of transmitted data packets in the channel, thus improving channel utilization. However, the disadvantage of SDRT is that, relying on only ACK, the sender cannot know which data packets are lost and how many packets need to be retransmitted, which can lead to redundant transmission and prolonged latency. Additionally, the calculation complexity of SDRT is high. Then, plenty of hybrid protocols based on ARQ are proposed to improve network performance. Reference [19] propose a hybrid ARQ scheme FOCAR, which integrates fountain coding and hop-by-hop ARQ to achieve high data delivery reliability. Mo *et al.* in [20] propose UW-HARQ, a hybrid ARQ protocol, which combines random binary linear coding-based FEC and ARQ (NACK and ACK) to reduce retransmission. Mo *et al.* then in [21] design CCRDT for underwater acoustic networks. CCRDT adopts random liner coding and multi-hop coordinated reliable data transfer scheme, which enhances packet recovery performance of SDRT. In [22], two hybrid ARQ schemes are proposed based on cross-layer separate parity transmission scheme. The two HARQ protocols improve throughput with less delay. [23] investigates two packet erasure coding schemes, including end-to-end and hop-by-hop to improve the data transmission reliability. In [24], a hybrid incremental redundancy ARQ protocol is proposed to improve the reliability of underwater acoustic links. In [25], TPNC is proposed as a reliable transport protocol based on two paths and network coding for UASNs. In TPNC, after twin paths being established, two groups of packets coded by network coding, are transmitted with their own shareable redundant packets over the two paths respectively to guarantee the data packet transmission reliability. In [26], an energy efficient data transport protocol based on network coding and hybrid automatic repeat request (NCHARQ) is proposed to ensure reliability and efficiency in UASNs. An adaptive window length estimation algorithm is also designed to optimize the throughput and energy consumption tradeoff. In addition, K. S. Geethu et al. in [27] also propose an erasure codes-based hybrid ARQ protocol that combines Reed-Solomon (RS) erasure codes-based FEC with a hop-by-hop selective retransmission scheme. Although these hybrid protocols based on ARQ can improve network performance effectively for reliable data communication, they introduce significant number of redundant packets transmissions and do not solve the problem of long data transmission delay in underwater DTN networks.

For adaptive cwnd-based transport layer protocol, ART-FEC protocol [28] is proposed for QoS-guaranteed image transmission in underwater wireless networks. ARTFEC is based on congestion window size control and the Q-learning optimal timeout selection mechanism. The Q-learning based optimal timeout selection algorithm can dynamically select the *RTO* timer according to the time-varying *RTT*, which improves the channel utilization efficiency and increases the network throughput. However, ARTFEC simply provides the selection sequence for the *RTO* to adapt to the *RTT* and does not solve the problem of long propagation delay in underwater acoustic communication.

To address the problems mentioned above, we propose an ORIT protocol for reliable data transfer in underwater DTN networks with extremely long propagation delays. The main idea of ORIT is that it is a hybrid of the optimal *RTO* timer and interval stop-and-wait transmission mechanism. For the optimal *RTO* timer, the *RTO* timer is reduced to shorten the delivery latency and improve the transmission efficiency in order to reduce the waiting time before retransmitting the lost data packets caused by the long *RTT*. For the interval stop-and-wait transmission mechanism, the retransmission begins under the premise of perceiving the previous transmission results to avoid data pseudo-retransmissions caused by the setting of the optimal *RTO* timer being shorter than the *RTT*, which can effectively improve the RRG performance in communication networks.

III. ORIT PROTOCOL

In this section, the ORIT execution rule is described and the theoretical model of ORIT is built up. According to the demands of different communication distances between the sender and the receiver in various environment conditions, ORIT can optimize the *RTO* timer to best RRG performance in underwater DTN networks. Considering the narrow

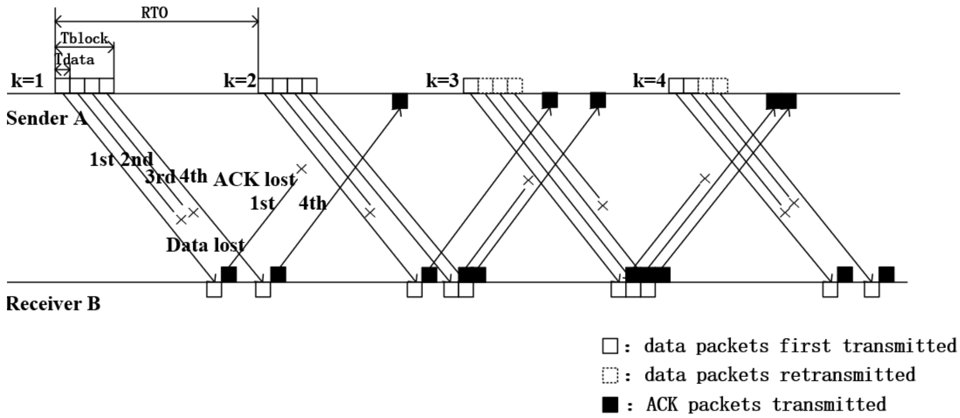

FIGURE 2. Information interaction process between sender A and receiver B of the ORIT protocol.

bandwidth characteristic of underwater acoustic channels, we take total data transmission traffic into account and explore the setting of the optimal *RTO* timer with the aim that the network achieves the best RRG.

A. ORIT EXECUTION RULE DESCRIPTION

The following is the description of the ORIT execution rule: Given the communication distance, two nodes adopt the peer-to-peer transport mode to exchange messages. The sender wishes to deliver data packets to the receiver with a certain confirmation mechanism to ensure that the receiver can receive messages correctly. Each time the sender sends data packets, the receiver takes a different action according to the receipt results. If the receiver receives the data packets correctly, it will provide feedback with the corresponding ACK information to the sender, ending the communication process. While the sender does not receive ACK when the *RTO* expires, it will retransmit the lost data packets. In each retransmission round-trip, data packets not yet transmitted from the file are added to the transmission window combined with the lost data packets to improve channel resource utilization. The interaction process continues until the receiver receives all of the data packets successfully. In ORIT, to avoid data pseudo-retransmissions because the *RTO* is set shorter than the *RTT*, the retransmission begins under the premise of having perceived the previous transmission results. Therefore, the retransmission is not oriented to last ACK, but rather to the ACK before last round-trip time.

Then, a specific information interaction process between sender A and receiver B of the ORIT protocol is shown in Fig. 2. Sender A divides the file received from the upper layer into several blocks with a fixed window size; each block contains several data packets. In Fig. 2, a single block comprises four data packets. Let L_{file}, L_{block}, L_{data} and L_{ack} denote the size of the file, block, data and ACK packet in bytes, respectively. In each transmission, sender A sends data packets that are not larger than L_{block}. In the first round-trip transmission, total data packets with size equal to L_{block} taken from the initial file are sent from sender A with the total transmission time T_{block}. During transmission, for example, the second and the third data packets are lost due to channel noise; thus, receiver B can only receive the first and the fourth data packets. Then, B sends two ACKs for the corresponding successfully transmitted data packets in response to A. However, the ACK for the first data packet is lost in the delivery process; thus, sender A can only receive one ACK for the fourth data packet. In the second round-trip transmission, sender A continues to send data packets with size equal to the L_{block} taken from the initial file because A does not know the results of the first transmission. This time, the second data packet and ACK for the third data packet are lost. Therefore, sender A receives only two ACKs for the first and the fourth data packets. In the third round-trip transmission, sender A has received one ACK for the fourth data packet from the first transmission. Thus, sender A takes one packet from the initial file and retransmits the first, second and third data packets lost in the first transmission round-trip. Additionally, the fourth data packet and ACK for the first data packet are lost in this round-trip transmission. Similarly, in the fourth round-trip transmission, according to the ACK packet feedback from the second transmission, sender A takes two packets from the initial file and retransmits the second and third data packets lost in the second round-trip transmission. Following this transmission mechanism, the underwater acoustic network implements data transfer.

B. THEORETICAL MODEL OF ORIT

In ORIT, the key factor is to find an optimal *RTO* timer for communication networks to achieve the best RRG performance. Let γ_n denote the best RRG performance we focused on; $RTO_{\min}$ and $RTO_{\max}$ denote the lower and upper limits of the *RTO* timer, respectively. The performance optimization problem can be expressed as:

$$\begin{cases} \max & \gamma_n(RTO) \\ s.t. & RTO_{\min} \leq RTO < RTO_{\max} \end{cases} \tag{1}$$

We then establish a theoretical model to solve the optimization problem as follows. Let p denote the bit error rate (BER) of underwater acoustic channels; given that all bits of data packets are transmitted independently, the loss probability of a data packet in bytes can be expressed as follows:

$$P_{data} = 1 - (1-p)^{8\times L_{data}} \tag{2}$$

Similarly, the loss probability of an ACK packet in bytes can be expressed as follows:

$$P_{ack} = 1 - (1-p)^{8\times L_{ack}} \tag{3}$$

Supposing the data transmission obeys the geometric probability distribution, the probability that a data packet is transmitted successfully from the sender to the destination in the kth transmission, represented as P_{data_kth}, can be calculated, where k denotes the times of transmission.

In the first transmission, P_{data_1st} can be expressed as:

$$P_{data\_1st} = 1 - P_{data} \tag{4}$$

In the second transmission, sender A does not know the number of data packets that are successfully transmitted or absent from in the first transmission. In this case, sender A does not retransmit the failed delivery data packets in the first transmission, but take the size of L_{block} data packets out from the file waiting to be sent in the queuing buffer of sender A to send. Therefore, the second transmission process is identical to the first, and P_{data_2nd} can be expressed as:

$$P_{data\_2nd} = P_{data\_1st} \tag{5}$$

In the third transmission, according to the rules for setting the *RTO* timer, sender A has received the ACK packet feedback from the first round-trip transmission. In this case, P_{data_3rd} can be expressed as:

$$P_{data\_3rd} = P_{data}\,(1 - P_{data}) \tag{6}$$

Similarly, the transmission process of the fourth transmission is identical to that of the third, and P_{data_4th} can be expressed as:

$$P_{data\_4th} = P_{data\_3rd} \tag{7}$$

Therefore, P_{data_kth} can be obtained by the following formula:

$$\begin{cases} P_{data\_kth} = P_{data}^{\frac{k-1}{2}}\,(1-P_{data}) & k = 2n-1 \\ P_{data\_kth} = P_{data\_(k-1)th} & k = 2n \end{cases} \tag{8}$$

where n is a positive integer and starts at 1 in the first transmission.

The number of data packets sent by sender A each time, defined as N_{data}, can be expressed as $N_{data} = L_{block}/L_{data}$. Correspondingly, in the kth transmission, the number of data packets that are successfully transmitted from the sender to the destination, denoted by N_{data_kth}, can be written as:

$$N_{data\_kth} = P_{data\_kth} \times N_{data} \tag{9}$$

Here, the number of failed and successful delivery data packets in the kth round-trip transmission can be obtained. Let P_{break} denote the probability of burst interrupt events in underwater acoustic communication, which obeys a normal distribution.

During the kth round-trip transmission, the number of data packets that failed to be transmitted due to the loss of ACK information and burst interrupt events, denoted by $N_{data\_n\_kth}$, can be calculated as follows:

$$N_{data\_n\_kth} = N_{data\_kth} \times (P_{ack}+P_{break}) \tag{10}$$

By contrast, the number of data packets that were successfully transmitted, denoted by $N_{data\_r\_kth}$, can be calculated as follows:

$$N_{data\_r\_kth} = N_{data\_old\_kth} + N_{data\_new\_kth} \tag{11}$$

In eq. (11), the first term contains two parts of the data packets. One is the data packets that were lost in the *(k-2)*th transmission for the first time, but successfully transmitted in the kth transmission. The other part is the data packets that were successfully transmitted in the kth transmission for the first time. $N_{data\_old\_kth}$ can be calculated as follows:

$$\begin{aligned} N_{data\_old\_kth} = {} & N_{data\_n\_(k-2)th} \times (1-P_{ack}) \times (1-P_{break}) \\ & + N_{data\_kth} \times (1-P_{ack}) \times (1-P_{break}) \end{aligned}$$

If sender A receives ACK of the data packets sent correspondingly, it will release the transmission buffer occupied by those data packets. Then, the region of transmission buffer released can be occupied by new data packets of the file taken out from the queuing buffer. Therefore, the second term represents the new data packets successfully transmitted which fill the remaining region of the transmission buffer with size of L_{block}. $N_{data\_new\_kth}$ can be calculated as follows:

$$\begin{aligned} N_{data\_new\_kth} = {} & \left(N_{data} - N_{data\_old\_kth}\right) \times (1-P_{data}) \\ & \times (1-P_{ack}) \times (1-P_{break}) \end{aligned}$$

The successful transmission of the entire file meets the condition, which can be written as:

$$\frac{L_{file}}{L_{data}} - \sum_{i=1}^{k} N_{data\_r\_kth} \ll 1 \tag{12}$$

According to the inequality eq. (12), the total number of times of transmission k can be obtained, then the total time required to complete the transmission from the sender to the destination of the entire file, denoted by T_{total}, can be calculated. Providing the round-trip time *RTT* of one-time transmission first can be represented as follows:

$$RTT = 2 \times T_{prop} + T_{data} + T_{ack} \tag{13}$$

In eq. (13), let T_{prop} denote the propagation time from the sender to the destination, formulated by $T_{prop} = L_{prop}/R_{prop}$, in which L_{prop} denotes the communication distance from the sender to the destination, and R_{prop} denotes the propagation speed, generally taking 1500 m/s. Let T_{data} denote the transmission time of data packets, formulated by $T_{data} = L_{data}/R_{data}$, in which R_{data} denotes the data packets transmission rate. Let T_{ack} denote the transmission time of the

ACK packet, formulated by $T_{ack} = L_{ack}/R_{ack}$, in which R_{ack} denotes the ACK packet transmission rate.

Here, the total time can be calculated by the following formula:

$$T_{total}{=}RTT_{last} + (k-1) \times RTO{+}T_{break} \tag{14}$$

where T_{break} is the duration of burst interrupt events and RTT_{last} is the round-trip time required for the last transmission, formulated as follows:

$$RTT_{last} = N_{data\_last} \times T_{data} + 2 \times T_{prop} + T_{ack}$$

where N_{data_last} denotes the data packets that need to be sent in the transmission buffer for the last round-trip transmission.

Ignoring the data processing time and queuing time, eq. (14) can well represent the total transmission time of the entire file. From eq. (14), it is obvious that the entire file transmission time is closely related to the *RTO* timer. Therefore, the appropriate setting of a shorter *RTO* timer is important for the reduction of underwater transmission delay.

According to the entire file transmission time, the goodput performance of ORIT for underwater communication networks, represented as γ, can be written as:

$$\gamma = \frac{L_{file}}{T_{total}} = \frac{L_{file}}{RTT_{last} + (k-1) \times RTO{+}T_{break}} \tag{15}$$

From eq. (15), a shorter *RTO* timer can lead to a better goodput performance. However, the shorter the *RTO* timer, the more data packets that flow through the channel, which causes heavy traffic in the communication network. Considering the narrow bandwidth characteristic of underwater acoustic channels, too many data packets will cause network congestion. This means that the setting of the *RTO* timer must be reasonable to ensure normal communication with the appropriate number of data packets that the channel can bear.

As mentioned above, the best goodput performance of the communication network with total data transmission traffic taken into account can be achieved under an optimal *RTO* timer. Then, with the *RTO* timer shorter than the *RTT*, the total number of data packets transmitted in the network with different round-trip transmissions are analyzed in the following.

When $k = 1$, the total number of data packets sent for the successful transmission of the entire file is expressed as:

$$N = \frac{RTT}{RTO} \times N_{data}$$

When $k = 2$,

$$N = \frac{RTT}{RTO} \times N_{data} + N_{data}$$

When $k = k - 1$,

$$N = \frac{RTT}{RTO} \times N_{data} + (k-2) \times N_{data}$$

To summarize the formula above, the total number of data packets can be obtained from the general formula:

$$N = \frac{RTT}{RTO} \times N_{data} + (k-2) \times N_{data} + N_{data\_last} \tag{16}$$

Then, the total number of data packets in bytes sent for the successful transmission of the entire file with transmission round-trip k can be written as:

$$N_{total} = N \times L_{data} \tag{17}$$

Considering the size of the file, the normalized total number of data packets sent is obtained through eq. (18), as shown at the bottom of this page.

Therefore, considering the normalized total number of data packets sent, the goodput performance of ORIT, which refers to the RRG, can be expressed as eq. (19), as shown at the bottom of this page.

According to the demand of different communication distances between the sender and the receiver in various environmental conditions, ORIT can obtain the optimal *RTO* timer under the best RRG performance in communication networks through the following relationship:

$$RTO{=}\ argmax\ (\gamma_n) \tag{20}$$

Here, the setting of the *RTO* timer is performed in the condition as follows:

$$\frac{\left(2 \times T_{prop}{+}T_{data} \times N_{data}{+}T_{ack}\right)}{2} \leq RTO < RTT \tag{21}$$

$$\begin{aligned} N_{total\_n} &= \frac{N_{total}}{L_{file}} {=} \frac{N \times L_{data}}{L_{file}} \\ &= \frac{\left(\frac{RTT}{RTO} \times N_{data} + (k-2) \times N_{data} + N_{data\_last}\right) \times L_{data}}{L_{file}} \end{aligned} \tag{18}$$

$$\begin{aligned} \gamma_n &= \frac{\gamma}{N_{total\_n}} \\ &= \frac{\frac{L_{file}}{RTT_{last}+(k-1)\times RTO+T_{break}}}{\frac{\left(\frac{RTT}{RTO}\times N_{data}+(k-2)\times N_{data}+N_{data\_last}\right)\times L_{data}}{L_{file}}} \\ &= \frac{{L_{file}}^2}{[RTT_{last} + (k-1) \times RTO{+}T_{break}] \times \left[\left(\frac{RTT}{RTO} \times N_{data} + (k-2) \times N_{data} + N_{data\_last}\right) \times L_{data}\right]} \end{aligned} \tag{19}$$

TABLE 1. Experiment paraments.

Experimental paraments	values
File size	6000, 10000, 14000 bytes
Block size	2000, 4000, 6000 bytes
Data packet size	100, 200, 300, 400, 500 bytes
ACK size	50 bytes
Transmission rate	10 kbps [30]
Break delay	N(0,1)
Break possibility	10^{-3}
BER	$10^{-4}, 1.5\times10^{-5}, 10^{-5}$
Propagation distance	3000, 4500, 6000 m
Experiment runs	100 trials

In the design of ORIT, $\frac{(2\times T_{prop}+T_{data}\times N_{data}+T_{ack})}{2} \le RTO$ is set to ensure that only the sender receives the ACK packet of the *(k-2)*th transmission, then the sender begins the *k*th transmission. If $RTO < \frac{(2\times T_{prop}+T_{data}\times N_{data}+T_{ack})}{2}$, the sender does not know the delivery results of the *(k-2)*th transmission, there will be pseudo-retransmitted data packets if the sender retransmits the data packets immediately. Thus, numerous pseudo-retransmissions can lead to congestion in narrow channels. Meanwhile, setting $RTO < RTT$ can reduce the long propagation delay.

Therefore, according to the optimal *RTO* timer in eq. (20), the best RRG performance in underwater DTN networks can be obtained. In next section, we focus on the relationship between the setting of the *RTO* timer and the performance of ORIT in underwater DTN networks.

IV. SIMULATIONS AND PERFORMANCE EVALUATIONS OF ORIT

A. SIMULATION SETTINGS

The mathematical modeling derived in the previous section for the performance of ORIT under various settings of the *RTO* timer is numerically experimented in this section. The experiment parameters are first listed in TABLE 1.

The file transmitted from the bundle layer to the transport layer numerically changes from 6000 to 14000 bytes, and RRG performance of ORIT is studied under various file sizes. Similarly, ORIT performance is also studied for different block and data packet sizes, which have varying sizes from 2000 to 6000 bytes and from 100 to 500 bytes, respectively. To verify the applicability of ORIT in various transmission delays, three propagation distances 3000, 4500 and 6000 m are set in actual communication scenarios. In different transport layer protocols, the *RTO* of ORIT is compared with other general *RTO* timers longer than the *RTT*. The rest of the specific parameter values are shown in TABLE 1.

B. NUMERICAL PRESENTATION AND PERFORMANCE EVALUATION OF ORIT

In this section, numerical experiments are presented to evaluate the performance of ORIT. We first explore the changes in the RRG performance with different settings of the *RTO* timer. Given the specific parameters, the RRG performances

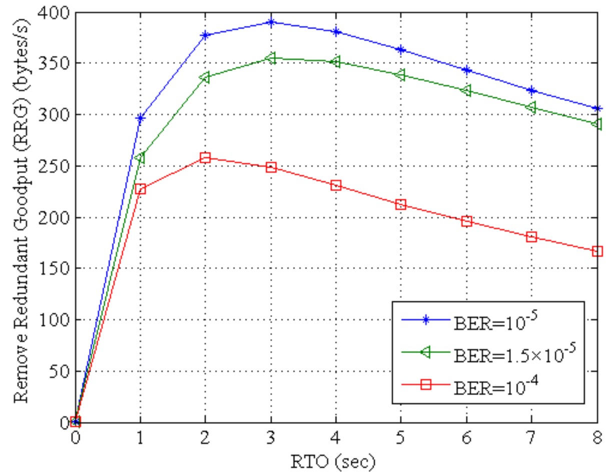

FIGURE 3. Remove redundant goodput performance various with the ***RTO*** timer under different channel BERs.

of ORIT are compared under different *RTO* settings, and the optimal *RTO* timer can be found for the best network performance. We then compare ORIT and other existing transport layer protocols with the *RTO* timer longer than the *RTT*.

1) REMOVE REDUNDANT GOODPUT PERFORMANCE OF ORIT

In Fig. 3, the relationship between the remove redundant goodput performance and the *RTO* timer under each of the three channel BERs, i.e., 10^{-5}, 1.5×10^{-5} and 10^{-4} is presented. The file delivered from the bundle Layer is 6000 bytes; additionally, the block size and the data packet size designed are 2000 bytes and 200 bytes, respectively. The propagation distance from the sender to the destination is 3000 m. The lower limit of the *RTO* timer in eq. (21), represented as the *RTOmin* is 2.82 s, and the *RTT* is 5.64 s. From Fig. 3, we can observe that under each of the three channel BERs, there exists an optimal *RTO* timer that allows the network to achieve the best RRG performance. This is reasonable because a longer *RTO* timer will cause an untimely retransmission of the lost data packets, and a shorter *RTO* timer will lead to numerous pseudo-retransmissions. We can observe that when the *RTO* timer approaches 0, the data packets transmit almost continuously, and large amounts of redundant data packets exist in the channel. The RRG performance approaches 0 at that moment. Therefore, too long or too short of an *RTO* timer can cause a decrease in protocol performance. Furthermore, as observed from Fig. 3, when the BER is 10^{-5}, the RRG performance is higher than the other two at BERs of 1.5×10^{-5} and 10^{-4} under the same *RTO* timer. Then, the protocol performance decreases with the increase of the BER. When the BER is 10^{-4}, the performance is the lowest among the three BERs under the same *RTO* timer. This is reasonable because with the increase of the BER, the probability of successful data transmission will decrease, which leads to additional retransmission times. Therefore, the larger BER increases the total file delivery time, reducing

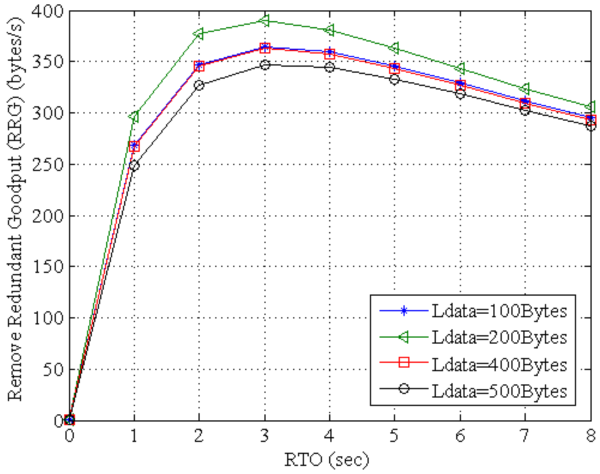

FIGURE 4. Remove redundant goodput performance various with the ***RTO*** timer under different data sizes.

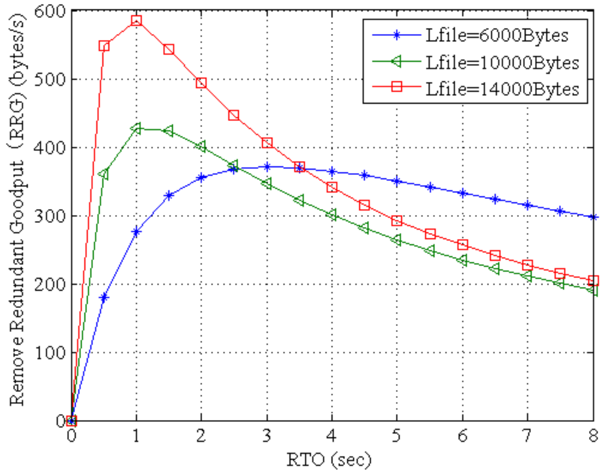

FIGURE 5. Remove redundant goodput performance various with the ***RTO*** timer under different file sizes.

the RRG performance. In addition, when the BER is smaller, 10^{-5} and 1.5×10^{-5} (as shown in Fig. 3), the optimal *RTO* timer under the best RRG performance, which is within the scope specified in eq. (21). When the BER is larger, i.e. 10^{-4}, the optimal *RTO* timer under the best RRG performance is less than the specified range in eq. (21). This is because when the BER is smaller, the protocol performance is relatively better, so a longer *RTO* timer can result in the communication network achieving the best performance. However, when the BER is larger, the protocol performance is relatively worse. Therefore, frequent retransmission is needed to offset the loss of the data in transmissions and cost for the best performance because of the larger BER. That is why the optimal *RTO* timer is short in this case, then the minimum threshold *RTOmin* in eq. (21) can be chosen as the actual optimal *RTO* timer.

Fig. 4 shows the relationship between the RRG performance and the *RTO* timer under each of the four data sizes, i.e., 100, 200, 400 and 500 bytes. The BER is 10^{-5}, and the remaining parameters are set the same as the others in Fig. 3. Additionally, the *RTOmin* is 2.82 s and the *RTT* is 5.64 s. From Fig. 4, we can observe that under each of the four data sizes, an optimal *RTO* timer exists to make the network achieve the best RRG performance. At the same time, when the data size is 200 bytes, the protocol performance is the best among all four data sizes. With the increase or decrease of data sizes, RRG performance is reduced. This is reasonable because when the data size is larger, there will be a long wait time before the data packets in the queuing buffer leave the sender, which will lead to low transmission efficiency. Furthermore, the probability of larger data packets being damaged by channel noises is higher; that is, the data transmission rate is lower, which leads to more retransmissions. Therefore, larger data packets lead to a decrease in the RRG performance. At the same time, when the data size is smaller, although the waiting time is short before data packets in the queuing buffer leave the sender, the next batch of data packets with the size of L_{block} to be transmitted must wait a long time until the next *RTO* timer expires, which can lead to a long propagation delay. Additionally, the smaller data packets will result in the transmission of too many packets in the narrow channel, which can cause many unnecessary collisions, also leading to a decrease in the RRG performance. Above all, an appropriate data size is significant for the improvement of the protocol performance.

Fig. 5 shows the relationship between the RRG performance and the *RTO* timer under each of the three file sizes, i.e., 6000, 10000 and 14000 bytes. The BER is 10^{-5} and the rest parameters are set the same as the others in Fig. 3. Additionally, the *RTOmin* is 2.82 s and the *RTT* is 5.64 s. Similar to the scenes analyzed previously, we can observe from Fig. 5 that under each of the three file sizes, an optimal *RTO* timer exists to result in the network achieving the best RRG performance. Furthermore, when the file size is 6000 bytes, the optimal *RTO* timer under the best RRG performance is within the scope specified in eq. (21). As the file size increases, the optimal *RTO* timer decreases. When file sizes are 10000 and 14000 bytes, the optimal *RTO* timer under the best RRG performance is less than the range specified in eq. (21). This is reasonable because when the file size is larger, there will be many initial data packets entering the transmission buffer in each retransmission, which leads to the decrease of the data transmission rate, and thus increases the transmission times. Then, frequent retransmission is needed to offset the transmission loss and to cost for the optimal protocol performance. Therefore, the optimal *RTO* timer is short in this case. That is also why when the file size is larger, the RRG performance drops rapidly under *RTO* timers outside the optimal *RTO* timer. In addition, due to the reliability of the protocol, the larger the file size, the higher the RRG performance. That is, the protocol performance under the optimal *RTO* timer improved with larger file size.

Finally, we observe the relationship between the RRG performance and the *RTO* timer under each of the three propagation distances, i.e., 3000, 4500 and 6000 m. The corresponding *RTOmins* are 2.82, 3.82 and 4.82 s, and the

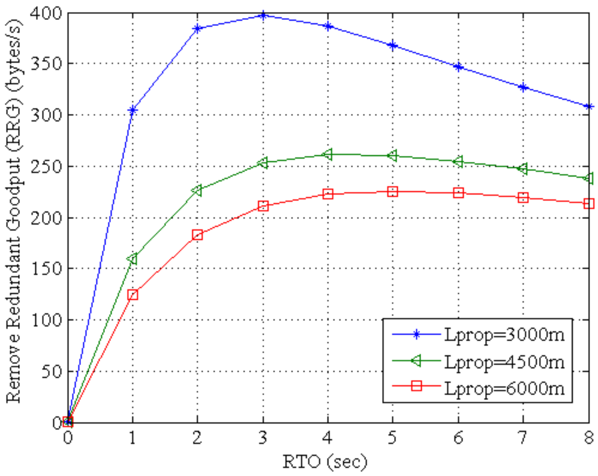

FIGURE 6. Remove redundant goodput performance various with the ***RTO*** timer under different propagation distances.

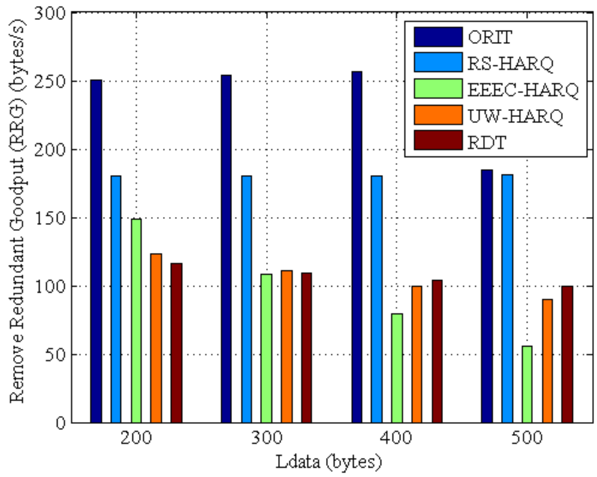

FIGURE 7. Remove redundant goodput performance comparison among ORIT, RS-HARQ, EEEC-HARQ, UW-HARQ and RDT under different data sizes.

TABLE 2. ***RTO*** **comparisons among ORIT, RS-HARQ, EEEC-HARQ, UW-HARQ and RDT.**

	ORIT	RS-HARQ	EEEC-HARQ	UW-HARQ	RDT
RTO (s)	2.82	9.26	10.82	11.50	7.65
RTT (s)	5.64	6.28	6.63	7.04	5.64

corresponding *RTTs* are 5.64, 7.64 and 9.64 s. From Fig. 6 we also find that under each of the three propagation distances, there is an optimal *RTO* timer resulting in the network achieving the best RRG performance. Therefore, according to the different distance requirements in actual communication, the optimal *RTO* timer can be set to achieve the best protocol performance.

In brief, regardless of the channel BER, data size, file size and propagation distance, there always exists an optimal *RTO* timer shorter than *RTT* to allow the underwater DTN network to achieve the best RRG performance. Thus, according to the actual communication environment, the best protocol performance can be obtained through the setting of the optimal *RTO* timer.

2) COMPARISON BETWEEN ORIT AND OTHER TRANSPORT LAYER PROTOCOLS

In this section, we compare ORIT and other existing transport layer protocols with the *RTO* timer longer than the *RTT* in DTN networks, i.e., RS-HARQ [27], EEEC-HARQ [23], UW-HARQ [20] and RDT [13]. Among these protocols, RS-HARQ, EEEC-HARQ and UW-HARQ combine the retransmission and encoding mechanism, where the two mechanisms do not affect each other; and RDT is based on the retransmission mechanism only.

Fig. 7 first presents the results of the RRG performance among ORIT, RS-HARQ, EEEC-HARQ, UW-HARQ and RDT with varying data sizes from 200 to 500 bytes. In ORIT, the *RTO* timer is shorter than the *RTT*, while in the others, the *RTO* timer is longer than the *RTT*. The results is provided with a setting of BER with 10^{-4} and with other parameters the same as the others in part A. From Fig. 7 we can observe that with different data sizes, the best RRG performance of ORIT is higher than those of the others. That is, ignoring the data size, the protocol performance of ORIT is always the best. In RS-HARQ, although the protocol has higher data delivery reliability, the transmission delay of significant number of encoded data packets can increase the file delivery time. At the same time, the propagation delays for data packets, ACK and NACK between the sender and the receiver are still long. In this scenario, the performance of ORIT with shorter *RTO* timer outperforms that of RS-HARQ with higher data delivery reliability but longer file delivery delay. In EEEC-HARQ, the sender has to send plenty of data packets because the amount of data to be sent by the relay nodes decrease in each hop. This results in longer transmission time and file delivery delay. In UW-HARQ, because of the lower encoding rate, the performance of it is relatively lower than that of RS-HARQ. In RDT, it has no data encoding process, thus the data transmission reliability is lower. Meanwhile, the file delivery delay of RDT is longer because of the general retransmission mechanism. All these reasons result in a lower performance of RDT. In the proposed ORIT scheme, there is no redundant encoded data packets to be sent and the fast retransmission mechanism for lost data packets can shorten the total file delivery time, thus optimizing the performance of the network. TABLE 2 gives the *RTT* and the corresponding *RTO* of ORIT, RS-HARQ, EEEC-HARQ, UW-HARQ and RDT, respectively. From TABLE 2 we observe that the *RTO* of ORIT with 2.82s is shorter than the *RTT* with 5.64s, while the *RTO* of the four other protocols are longer than their own *RTT*. This fact further demonstrates the importance of shorter retransmission time for improving the network performance.

In Fig. 8, the comparison of the RRG performance among ORIT, RS-HARQ, EEEC-HARQ, UW-HARQ and RDT with three different block sizes, i.e., 2000, 4000 and 6000 bytes

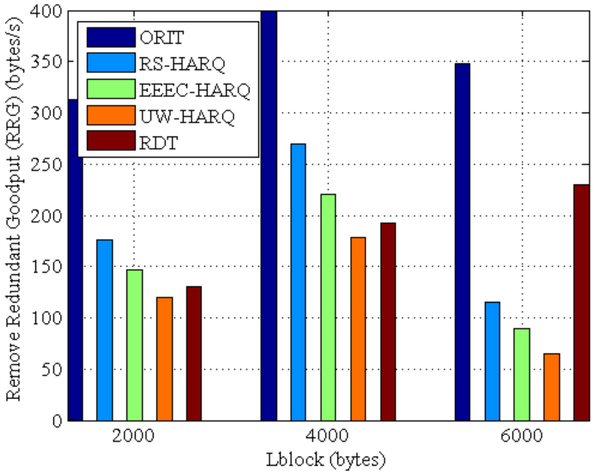

FIGURE 8. Remove redundant goodput performance comparison among ORIT, RS-HARQ, EEEC-HARQ, UW-HARQ and RDT under different block sizes.

is presented. The file size is set at 12000 bytes to satisfy the changes of the block size, and the BER is set at 10^{-4}. Additionally, other parameters are the same as those in part A. From Fig. 8 we can observe that each protocol has an optimal block size to achieve its' own best performance. Moreover, with different block sizes, the best RRG performance of ORIT is higher than those of the others. That is, ignoring the block size, the protocol performance of ORIT is always the best. The reason is the same as the explanation in Fig. 7, that there is no redundant encoded data packets to be sent and the fast retransmission mechanism for lost data packets can shorten the total file delivery time, thus optimizing the performance of the network.

Finally, Fig. 9 presents the comparison of the RRG and average end to end delay performance among ORIT, RS-HARQ, EEEC-HARQ, UW-HARQ and RDT with three different propagation distances, i.e., 3000, 4500 and 6000 m. Average end to end delay is defined as the average time taken by transmitting the entire file successfully from source to sink in multiple experiments, including propagation time of data and ACK, data transmission time, retransmission time and interrupt time. The BER is set at 10^{-4} and other parameters are the same as those in part A. From Fig. 9(a) we can observe that with different propagation distances, the best RRG performance of ORIT is higher than those of the others. That is, ignoring the propagation distance, the protocol performance of ORIT is always the best. This is also the same reason that there is no redundant encoded data packets to be sent and the fast retransmission mechanism for lost data packets can shorten the total file delivery time, thus optimizing the performance of the network. Meanwhile, with the increase in the propagation distance, the network performance differences among five protocols decreases. This implies that the negative impact of the long propagation distance weakens the optimization effect of the protocol performance. Fig. 9(b) shows the relationship between average end to end delay and propagation distance. With the increase of propagation

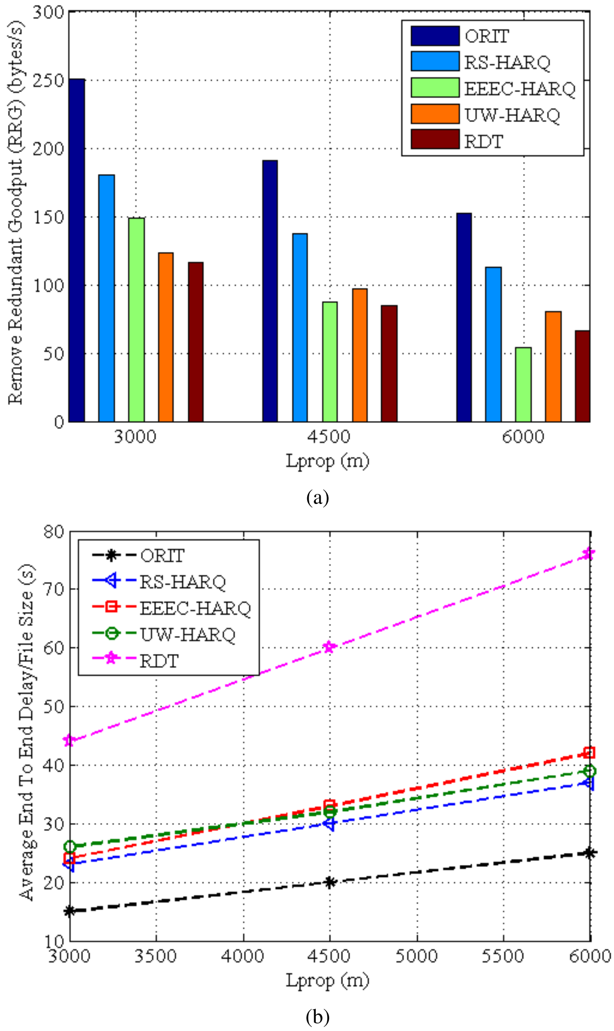

FIGURE 9. (a) Remove redundant goodput performance comparison among ORIT, RS-HARQ, EEEC-HARQ, UW-HARQ and RDT under different propagation distances. (b) Average end to end delay performance comparison among ORIT, RS-HARQ, EEEC-HARQ, UW-HARQ and RDT under different propagation distances.

distance, the average end to end delay increases. We can see that under each of the three propagation distances, ORIT always takes minimal time to delivery the total file with fast retransmission mechanism. RS-HARQ, EEEC-HARQ and UW-HARQ take longer time due to the transmission of redundant encoded data packets and general longer *RTO* timer. For RDT, because of the low data transmission reliability and long data retransmission time, it has to retransmit more lost data packets and takes a longer time to complete the total file delivery mission.

In conclusion, in underwater communication environment, the data delivery efficiency of ORIT with an optimal *RTO* timer shorter than the *RTT* is better than those of other existing transport layer protocols with an *RTO* timer longer than the *RTT* even with encoding schemes in DTN networks. The numerical results of the RRG performance, which consider the characteristic of narrow bandwidth in underwater acoustic channels, make the experiment results more convincing.

V. CONCLUSION

In this paper, a transport layer protocol, ORIT, is proposed for the DTN structure applied in underwater acoustic communication networks with long propagation delay, narrow channel bandwidth and frequently disconnected interruption. In ORIT, the optimal *RTO* timer within a reduced *RTO* timer range is set, which can effectively shorten the transmission time in underwater environments and obviously improve the data delivery efficiency. To avoid data pseudo-retransmissions caused by the setting of the *RTO* timer shorter than the *RTT* in narrow channel bandwidth, an interval stop-and-wait transmission mechanism is adopted. Numerical experiments are presented to evaluate the performance of ORIT with different settings of the *RTO* timer. Considering the total data packets transmitted in the channel, we use RRG to evaluate the protocol performance. Regardless of the channel BER, data size, file size and propagation distance, an optimal *RTO* timer shorter than the *RTT* always exists to allow underwater DTN networks to achieve the best RRG performance. We compare ORIT with other transport layer protocols, i.e., RS-HARQ, EEEC-HARQ, UW-HARQ and RDT, and found that in underwater communication environment, the performance of ORIT with an optimal *RTO* timer shorter than the *RTT* is better than those of other existing transport layer protocols with the *RTO* timer set longer than the *RTT* in DTN networks even have encoding schemes. That is in our proposed protocol with the optimal *RTO* timer scheme, ORIT can achieve a superior RRG performance relative to other general schemes.

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