Reliable Transmission Protocol for Underwater Acoustic Networks

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Abstract Underwater Acoustic Networks (UANs) use acoustic communication and are characterized by limited bandwidth capacity, high energy consumption, long propagation delay, which cause the traditional protocols designed for radio channels to be either inapplicable or to be inefficient for UANs. The chapter introduces a three-layer protocol architecture for UANs which is Micro-ANP (including Application, Network-transport and Physical layer). Further, based on the Micro-ANP architecture and Recursive LT (RLT) code, a handshake-free reliable transmission mechanism is presented in detail.

1 Challenges of UANs

Recently, Underwater Acoustic Networks (UANs) research has attracted significant attention due to the potential for applying UANs in environmental monitoring, resource investigation, disaster prevention, and so on [1-10]. UANs use acoustic communication, but the acoustic channel is characterized by high bit errors (on the order of magnitude of 10^{-3}-10^{-7}), long propagation delay (at a magnitude of a few seconds), and narrow bandwidth (only scores of kbps). The result is that the terrestrial-based communication protocols are either inapplicable or inefficient for UANs. Compared with conventional modems, the acoustic modems used in UANs consume more energy. However, the nodes are battery-powered and it is considerably more difficult to recharge or replace nodes in harsh underwater environments. Furthermore, underwater nodes are usually deployed sparsely, move passively with water currents or other underwater activity, and some nodes will fail due to energy depletion or hardware faults; therefore the network topology of UANs usually changes dynamically, which causes significant challenges in designing protocols for UANs.

Applications of UANs in areas such as business, scientific research and military are usually sensitive: outsiders are not allowed to access the sensitive information, and anonymous secure communication is broadly applied. However, thus far, to the best of our knowledge, there are few papers concerning secure communications protocols for UANs [11-14]. The nature of opening and sharing of underwater acoustic channel makes communications inherently vulnerable to eavesdropping and interference. Because of the highly dynamic nature of UANs, as well their lack of centralized management and control, designing secure routing protocols that support anonymity and location privacy is a large challenge.

In UANs with dynamic topology and impaired channel, network efficiency following the traditional five-layered architecture was obtained by cross-layer designs, which cause numerous complicated issues that are difficult to overcome. The chapter introduces a three-layer protocol architecture for UANs, which includes application layer, network-transport layer and physical layer and is named Micro-ANP. Based on the three-layer Micro-ANP architecture, the chapter provides a handshake-free Media Access Control (MAC) protocol for UANs, and achieves reliable hop-by-hop transmissions.

The remainder of the chapter is organized as follows. Section 2 presents the Micro-ANP architecture. Section 3 reviews the research on reliable transmission mechanism so far. Section 4 details the handshake-free reliable transmission protocol for UANs based on Micro-ANP architecture and RLT code. Section 5 makes a conclusion and has a discussion about new trends of UANs research.

2 Micro-ANP Architecture

The majority of research on UANs has focused primarily on routing or MAC protocols, and few studies have investigated protocol architecture for UANs. The energy, computation and storage resources of UANs are seriously constrained; consequently, the protocol stack running on UANs nodes should not be complicated. However, most research on UANs so far has followed the traditional five-layered architecture in network design, and in tough condition such as dynamic topology, seriously impaired channel and scarce resources, network efficiency was obtained by cross-layer designs, which cause numerous complicated issues that are difficult to overcome. UANs need a simple and efficient protocol architecture. Du *et al.* provided a three-layered Micro-ANP architecture for UANs, which is composed of an application layer, a network transport layer and a physical layer as well as an integrated management platform, as shown in Fig. 2.1 [15].

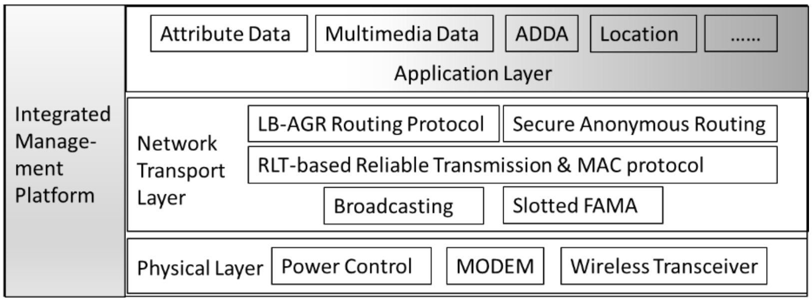

Fig. 2.1. Micro-ANP Architecture

The network transport layer in Micro-ANP is primarily responsible for reliable hop-by-hop transmission, routing, and channel access control. In Micro-ANP, broadcasting, Level-Based Adaptive Geo-Routing (LB-AGR) and a secure anonymous routing are the three major routing protocols that are applicable to dynamic underwater topology[7] [16]. A secure anonymous routing protocol can achieve anonymous communication between intermediate nodes as well as two-way authentication between source and destination nodes without any real-time online Public Key Generator (PKG), thus decreases the network delay while improving network scalability. In Micro-ANP, slotted Floor Acquisition Multiple Access (slottedFAMA) and a RLT Code based Handshake-Free (RCHF) reliable MAC protocol are the two-channel access control mechanism [9] [17].

Micro-ANP is a three-layered architecture that allows intermediate nodes to perform Application Dependent Data Aggregation (ADDA) at the application layer. Without requiring a cross-layer design, Micro-ANP can make efficient use of scarce resources. Moreover, Micro-ANP eliminates inapplicable layers and excessive repeated fields such as address, ID, length, Frame Check Sequence (FCS), and so on, thus reducing superfluous overhead and energy consumption. The head fields of the network transport layer are listed in Table 2.1.

The application priority field is used to distinguish between different applications as shown in Table 2.2. This is because different applications have different priorities and require different Quality of Service (QoS) and their messages are transmitted using different routing decisions. Other fields in Table 2.2 will be explained in the respective protocol overview of the network transport layer.

From Table 2.1, we can see that the common head-length of Micro-ANP is less than 20 bytes. In comparison, the total head-length of well-known five-layer models is more than 50 bytes. Therefore, Micro-ANP protocol greatly improves data transmission efficiency.

Table 2.1 Head Fields of Micro-ANP

Bits: 8	8	8	2	6	1	1	24	8

level of sender	sender ID	receiver ID	type 00: Data 01: Ack 10: Control	frame sequence number	immediately ack 1: yes 0: no	If block 1: Yes 0: No	IDs of original Packets	block ID
Bits:6	1	2	1	48	4	8	variable	16
block size	direction 0: down 1: up	Sink ID	(source| destination) 0: position 1: node ID	(source|destination) position or ID Full “1” for broadcast	application priority (application type)	load length	data	FCS

Table 2.2 Application Priority

Priority	Upper Protocol	Priority	Upper Protocol
0	Attribute data	4	Video
1	Integrated management	5	Emergency alarm
2	Image	6	
3	Audio	7	

3 Overview of Reliable Transmission Mechanism

Considering the challenges for UANs, the existing solutions of terrestrial Radio Frequency (RF) networks cannot be applied directly to UANs, regardless of the MAC mechanism used, the reliability of data transmission, or the routing protocol. Sustained research work over the last decade has introduced new and efficient techniques for sensing and monitoring marine environments; several issues still remain unexplored. The inapplicability of conventional reliable transport mechanisms in UANs is analyzed as follows.

1) The high bit error rates of acoustic channels lead to high probability of packet erasure and a low probability of success in hop-by-hop transfers. Therefore, traditional end-to-end reliable transport mechanisms may incur too many re-transmissions and experience too many collisions, thus reducing channel utilization.

2) The low propagation speed of acoustic signals leads to long end-to-end delays, which causes issues when controlling transmissions between two end-nodes in a timely manner.

3) The Automatic Repeat Request (ARQ) mechanism re-transmits lost packets, but it requires an ACK (acknowledgement) for packets received successfully. It is well-known that the channel utilization of the simple stop-and-wait ARQ protocol is very low in UANs due to long propagation delays and low bit rates. In addition, acoustic modems adopt half-duplex communication, which limits the choices for

efficient pipelined ARQ protocols. Even worse, if the ACKs are lost, the successfully received packets will be re-transmitted by the sender, further increasing the bandwidth and energy consumed.

Some reliable transport protocols resort to Forward-Error-Correcting (FEC) to overcome the inherent problems with ACKs. FEC adopts erasure codes and redundancy bits. The payload bits of FEC are fixed prior to transmission. Before transmitting, the sender encodes a set of n original packets into a set of N $(N \geq n)$ encoded packets. Let$m = N - n$, and m redundant packets are generated. To reconstruct the n original packets, the receiver must receive a certain number (larger than n) of encoded packets. The stretch factor is defined as N/n, which is a constant that depends on the erasure probability. However, the error probability of UANs channels is dynamic; overestimated error probability will incur additional overhead and underestimated error probability will lead to transmission failure.

Reed and Solomon proposed the Reed-Solomon code based on some practical erasure codes [18]. Reed-Solomon code is efficient for small n and m values. However, the encoding and decoding algorithms require field operations, resulting in a high computation overhead that is unsuitable for UANs due to the nodes' limited computational capabilities. Luby et al. studied a practical Tornado code which involves only XOR operations [19]. In addition, the encoding and decoding algorithms are faster than those used for Reed-Solomon code. However, the Tornado code uses a multi-layer bipartite graph to encode and decode packets, resulting in a high computation and communication overhead for UANs. Xie et al. presented an Segmented Data Reliable Transfer (SDRT) protocol [20]. SDRT adopts Simple Variant of Tornado (SVT) code to improve the encoding/decoding efficiency. Nevertheless, after pumping the packets within a window into the channel quickly, the sender sends the packets outside the window at a very slow rate until it receives a positive feedback from the receiver, which reduces channel utilization. Mo et al. investigated a multi-hop coordinated protocol for UANs based on the GF(256) random-linear-code to guarantee reliability and efficiency [21]. However, the encoding vectors are generated randomly; consequently, the probability of successfully recovering K data packets from K encoded packets could not be guaranteed. Moreover, the decoding complexity was higher than other sparse codes. Furthermore, the multi-hop coordination mechanism requires time synchronization and is restricted to a string topology in which there is a single sender and a single receiver.

Digital fountain codes are sparse codes on bipartite graphs that have high performance [21, 23]. They are rate-less, i.e., the amount of redundancy is not fixed prior to transmission and can be determined on the fly as the error recovery algorithm evolves. These codes are known to be asymptotically near-optimal for every erasure channel, and they allow for lightweight encoder and decoder implementations. Luby proposed the LT code, in which the decoder is capable of recovering the original symbols at a high probability from any set of output symbols whose size is close to the originals [24]. However, the LT code was designed for large numbers of data packets, which is not typically the case in UANs—especially for mobile networks where the transmission time between two nodes is very limited because of node mobility. Furthermore, the degree distribution used in LT code results in a large number of nodes in the graph, causing a large overhead for each packet.

4 Reliable Transmission Protocol for UANs

In this section, based on digital fountain code, a Recursive LT (RLT) code with a small degree distribution is proposed along with a reliable and handshake-free MAC protocol called as RCHF MAC protocol.

4.1 RLT Code

The coding scheme can greatly impact system performance. In this section, we present a Recursive LT (RLT) code, which achieves fast encoding and decoding. Given that packet loss is independent, we use a bipartite graph $G = (V, E)$ with two levels to represent the RLT code, where E is the set of edges and V is the set of nodes in the graph. $V = D \cup C$, where D is the set of input packets and C is the set of encoded packets. The edges connect the nodes in D and C.

(1) Encoding

Consider a set of $k$input (original) packets, each having a length of l bits. The RLT encoder takes k input packets and can generate a potentially infinite sequence of encoded packets. Each encoded packet is computed independently of the others. More precisely, given k input packets $\{x_1, x_2, \cdots, x_k\}$ and a suitable probability distribution $\Omega(d)$, a sequence of encoded ets $\{y_1, y_2, \cdots, y_j, \cdots, y_n\}, n \geq k$, are generated as shown in Fig. 4.1. The parameter d is the degree of the encoded packets-the number of input packets used to generate the encoded packets and $d \in \{1, 2, \cdots, k\}$ (e.g., the degree of packet y_2 is 2 while the degree of packet y_8 is 3 in Fig. 4.1).

To restore all the k original packets at the receiver, the number of encoded packets received successfully is subject to be greater than k . Let $\mathrm{n} = (k + \xi)/(1 - P_p)$; here, P_p is the erasure probability of an underwater acoustic channel (i.e., the PER), and $\xi (\xi > 0)$ corresponds to the expected number of redundant encoded packets received. The ξ redundant packets are used to decrease the probability that the receiver fails to restore the original k input packets in only one transmission phase. The sequence of encoded packets is $y_1, y_2, \cdots, y_j, \cdots, y_n \in C$. The RLT encoding procedure is as follows.

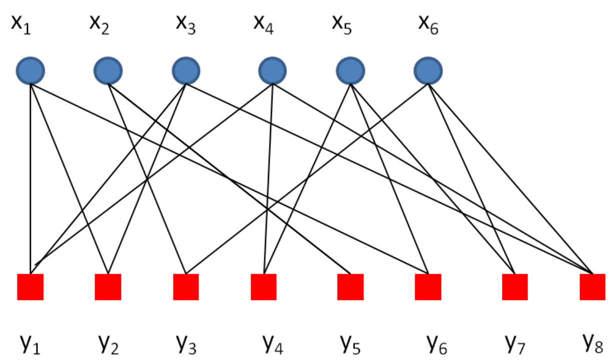

Fig. 4.1. Encoding graph of RLT code

1) From D, the set of input packets, successively XOR the k packets to generate one encoded packet with degree k, then duplicate the packet to obtain $\lceil 1/(1-P_p) \rceil$ copies.

2) From set D, select $\lceil m/(1-P_p) \rceil$ distinct packets randomly to constitute a seed set S_1, and generate $\lceil m/(1-P_p) \rceil$ encoded packets with degree one. Here, m is the expected number of encoded packets received successfully with degree one. In reality, we can set $1 \leq \mathrm{m} \leq \max(\lfloor k/4 \rfloor, 1)$.

3) Let $S_2 = D - S_1$. From the set S_2, uniformly select $\lceil k/(2(1-P_p)) \rceil$ input packets at random, and perform the XOR operation, randomly selecting one packet in the set S_1 to generate $\lceil k/(2(1-P_p)) \rceil$ encoded packets with degree two.

4) Let $S_3 = D - S_1 - S_2$. If S_3 is not null, select $\lceil k/(6(1-P_p)) \rceil$ input packets at random from set S_3; otherwise, from set D, perform the XOR operation using one packet from S_2 and another from S_1 to generate $\lceil k/(6(1-P_p)) \rceil$ encoded packets with degree three.

5) Let $S_4 = D - S_1 - S_2 - S_3$. If S_4 is not null, randomly select $\lceil (\xi + k/3 - m - 1)/(1-P_p) \rceil$ input packets from set S_4; otherwise, from set D, perform the XOR operation using three packets from $S_1, S_2,$ and S_3, respectively, to generate $\lceil (\xi + k/3 - m - 1)/(1-P_p) \rceil$ encoded packets with degree four.

(2) Decoding

When an encoded packet is transmitted over an erasure channel, it is either received successfully or lost. The RLT decoder tries to recover the original input packets from the set of encoded packets received successfully. The decoding process of RLT is as follows.

1) Find an encoded packet y_j which is connected to only one input packet x_i. If the receiving node fails to find any such encoded packet, stop decoding.

2) Set $x_i = y_j$.

3) Set $y_m = y_m \oplus x_i$ for each encoded packet which is connected to x_i, denoted by y_m. Here, $\oplus$ indicates the XOR operation.

4) Remove all the edges connected to x_i.

5) Go to Step 1.

(3) Degree Distribution

The limited delivery time between two nodes caused by node mobility leads to the constraint that digital fountain codes must work with small k values in UANs communications. In RLT, to reconstruct the input packets, the degree distribution of the received encoded packets should have the following properties.

1) The received encoded packets should connect all the input packets.

2) The process of encoding and decoding should not involve too many XOR operations.

3) At least one encoded packet with degree one should be successfully received by the receiver.

Given the high bit error, P_b, which is on the order of magnitude of 10^{-3}-10^{-7}, the PER, P_p, is given by Eq. (4.1):

$$p_p = 1-(1-p_b)^l \quad , \tag{4.1}$$

where l is the packet size. As discussed earlier, in Micro-ANP architecture, the optimal packet size is greater than one hundred bytes, and P_p is non-negligible in Eq. (4.1). Considering the k input packets, to address the properties of degree distribution discussed above, the degree distribution of the encoded packets in the sending nodes is given by Eq. (4.2):

$$\Omega(d) = \begin{cases} \dfrac{m}{\xi+k}, & d=1; \\ \dfrac{k}{d(d-1)(\xi+k)}, & d=2,3; \\ \dfrac{\xi+(1/3)k-(m+1)}{\xi+k}, & d=4; \\ \dfrac{1}{\xi+k}, & d=k; \end{cases} \tag{4.2}$$

where $\sum_d \Omega(d) = 1$.

Lemma 1. *The average degree of* encoded packets $\lambda \approx 3.7$.

Proof. From the degree distribution given by Eq. (4.2), we obtain:

$$\lambda = E(d) = \sum_{d=1}^{4} (d \times \Omega(d))$$

$$= \frac{1 \times m}{\xi + k} + \frac{2 \times k}{2 \times 1 \times (\xi + k)} + \frac{3 \times k}{3 \times 2 \times (\xi + k)} + \frac{4 \times (\xi + 1/3k - (m+1))}{\xi + k} + \frac{k}{\xi + k}$$

$$= 3\frac{2}{3} + \frac{\frac{\xi}{3} - 3m - 4}{\xi + k}.$$

Usually, $|(\xi/3) - 3m - 4| \ll |\xi + k|$, so $\lambda \approx 3\frac{2}{3} \approx 3.7$.

Given the block size k, from Lemma 1, we can derive the decoding complexity of RLT is about $3.7k$ which is linear to the number of input packets. A comparison of the encoding/decoding complexity of various codes is shown in Table 4.1.

Table 4.1. Decoding complexity comparison

Code	Encoding/Decoding Complexity
GF (256) in [21]	$O(k^3)$
LT	$k \ln_e^k$
SDRT in [20]	$k \cdot \ln(1/\varepsilon)$
RS	$k(N-k)\log_2^N$
RLT	$3.7k$

In this section, based on the digital fountain code, we propose a recursive LT (RLT) code with small degree distribution, and introduce the erasure probability of channel P_p into the RLT code for the first time to improve the decoding probability at the receiving node. RLT is applicable to dynamic UANs with limited transmission time between two nodes; it reduces the overhead of encoding and decoding and substantially improves the efficiency of decoding process.

4.2 RCHF: RLT Code based Handshake-Free Reliable Transmission Protocol

After solving the problems of degree distribution, encoding and decoding of RLT in advance, a reliable RLT-based media access control protocol should be presented that nodes can use to communicate in real time. Wireless transceivers usually work in half-duplex mode: a sending node equipped with a single channel is unable to receive packets while it is transmitting; therefore, the RCHF solution is supposed to avoid interference caused by transmitting to a node in a sending state. So far, in MAC solutions of wireless multi-hop packet networks, an RTS/CTS handshake is used to dynamically determine whether the intended receiver is ready to receive a frame. For underwater sensors, the rate at which data bits can be generated is approximately 1-5 bps and the optimal packet-load for UANs is about one hundred bytes. In contrast, the length of an RTS frame is a few dozen bytes. Therefore, RTS/CTS frames are not particularly small compared with data frames; consequently, the benefits from using RTS/CTS handshake are unremarkable. Moreover, considering the characteristics of acoustic communication (i.e., low bandwidth, long propagation delay, etc.), RTS/CTS handshake decreases channel utilization and network throughput dramatically while prolonging end-to-end delay. Therefore, coupled closely with the RLT code, we propose a RCHF protocol which is a state-based handshake-free reliable MAC solution for UANs.

4.2.1 Reliable Transmission Mechanism

In the RCHF MAC solution, a source node first groups input packets into blocks of size k (i.e., there are k input packets in a block). Then, the source node encodes the k packets, and sends the encoded packets to the next hop. When k is equal to 50, the minimum time interval for transmitting a block between two neighbor nodes is approximately 60 s, which is in compliance with the requirements of the limited transmission time between two neighbor nodes in dynamic UANs. By setting the block size k appropriately, RCHF can control the transmission time, allowing the receiver to be able to receive sufficient encoded packets to reconstruct the original block even when the nodes are moving. Application data are transferred from a source node to a sink node block by block and each block is forwarded via RLT coding hop-by-hop.

In the RCHF protocol, a node sending packets is considered to be in the transmission phase. To facilitate receiving an ACK for transmitted packets, avoid conflicts between transmitting and receiving, and compromise between transmission efficiency and fairness, two transmission constraints are defined as follows.

1) The maximum number of data frames allowed to be transmitted in one transmission phase is N_{max}.

2) The minimum time interval between two tandem transmission phases of the same node is T_a. The node waiting for T_a expiration is considered to be in a send-avoidance phase. At present, underwater acoustic modems are half-duplex, the delay for state transition between sending and receiving usually ranges from hundreds of milliseconds to several seconds, which is close to the magnitude of the maximum round-trip time (RTT) [18]. Therefore, to facilitate the receiver to switch to the sending state to transmit the ACK, we set $T_a = 2 \times RTT$.

After transmitting N ($N \leq N_{max}$) encoded packets, the sender switches to the receiving state and waits for the receiver's ACK. To have a high probability of being able to reconstruct the original k input packets at the receiver, the number of encoded packets received successfully is supposed to be larger than k, denoted as $k + \xi$. Considering the high packet error rate, P_p, we set $N = (k + \xi)/(1 - P_p)$. The parameter ξ, ($\xi > 0$) is fixed and corresponds to the expected number of redundant encoded packets the receiver will receive. The ξ redundant packets are used to decrease the probability that the receiver fails to restore the original k input packets in the transmission phase, and the factor $1/(1 - P_p)$ is used to compensate for channel errors.

The ACK frame includes the number of frames received at the receiver as well as the indices of unrecovered input packets. The number of frames received successfully can be used to update the packet error rate P_p on the fly. If the receiver can reconstruct the whole block, it sends back an ACK with "null" in the index field.

Given k_1 input packets unrecovered after the previous transmission phase, the sender encodes and transmits N_1 encoded packets with the degree distribution given by Eq. (4.2) in which k is replaced by k_1. $N_1 = (k_1 + \xi)/(1 - P_p)$. Then the sender collects the feedback from the receiver again. This process repeats until the sender receives an ACK with "null" in the index field.

4.2.2 State-Based Handshake-Free Media Access Control

After network initialization, each node maintains one dynamic neighbor table that includes a state field containing the real-time state of neighbor nodes as shown in Table 4.2. Here, state "0" indicates that the neighbor node is in sending state, state "1" indicates that the neighbor node is receiving frames from other nodes, "2" denotes an unknown state, and "3" means the neighbor node is in the send-avoidance phase.

Table 4.2. The state table of neighbor nodes

Value	State
0	sending state
1	receiving frame from other nodes
2	unknown state
3	transmission-avoidance

The format of frames in our protocol is shown in Table 4.3. The level field contains the forwarder's level, the frame sequence number is used to identify the frame in one frame-sequence during one transmission phase, the original packet ID field is used to indicate the IDs of packets that are XORed, and the immediate

ACK field is used to inform the receiver whether to return an ACK immediately, where “1” means "yes" and “0” means "no." The first nine bytes are used by the RCHF MAC protocol to realize reliable transmission hop-by-hop; the fields are updated hop-by-hop. The fields from the tenth to the sixteenth bytes are used by the LB-AGR routing protocol and are omitted here for simplicity.

Table 4.3. The format of data frame

Bits: 8	8	8	2	6	1	1
level of sender	Sender ID	Receiver ID	type 00: Data 01: Ack 10: Control	frame sequence number	immediately ack 1: yes 0: no	If block 1: Yes 0: No
Bits:24	8	6	…	8		Variable
IDs of original Packets	block ID	block size	…	load length		data

When a node has packets to send, it searches the neighbor table for the state field of the intended receiver. If the state is “0” or “1”, it will delay delivery until the state is greater than one; otherwise, the node becomes a sender, switches into the transmission phase and starts to deliver frames. The pseudocode for sending packets is omitted.

4.3 Simulation Result of RCHF

In this section, we evaluate the performance of the RCHF protocol by simulation experiments. All simulations are performed using Network Simulator 2 (NS2) with an underwater sensor network simulation package extension (Aqua-Sim). Our simulation scenario is similar to reality; one hundred nodes are distributed randomly in an area of 7000 m×7000 m×2000 m. The simulation parameters are listed in Table 4.4.

The protocol is evaluated in terms of average end-to-end delay, end-to-end delivery ratio, energy-consumption and throughput. We define the delivery ratio and throughput of the RCHF protocol as follows.

1) The end-to-end delivery ratio is defined by Eq. (4.3):

$$end\text{-}to\text{-}end\ delivery\ ratio = \frac{\#\ \text{of}\ packets\ received\ successfully\ at\ sink}{\#\ of\ packets\ generated\ at\ sources} \quad (4.3)$$

2) The throughput is defined as the number of bits delivered to the sink node per second (bps)

Table 4.4. Simulation parameters

Parameter	Value
Block Size k	50
Packet Length l	160 bytes
Bandwidth	10 kbps
Routing protocol	static
Traffic	CBR
Transmission Range	1500 m
MAC Protocol	802.11

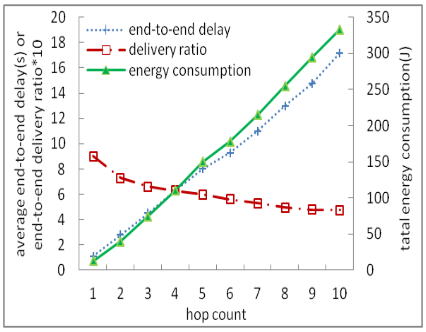

Fig. 4.2. Performance vs. hop count

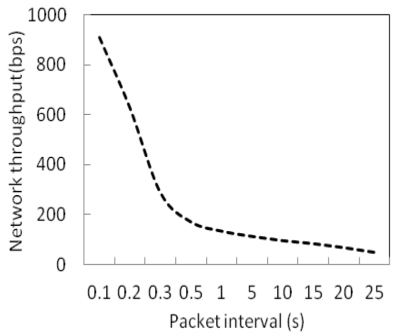

Fig. 4.3. Throughput vs. packet interval(s)

As shown in Fig. 4.2, the end-to-end delivery ratio of the RCHF protocol is close to “1” when the hop count is “1” and decreases slightly as the hop count increases, which is considered good performance for UANs from a delivery ratio aspect. Fig. 4.2 also shows that the end-to-end delay and total energy-consumption rise with the hop count which is understandable. Note that the real value of the end-to-end delivery ratio is the value of the ordinate axis divided by 10.

As shown in Fig. 4.3, the network throughput of RCHF decreases as the interval time between two successive packets generated by the source node increases. This occurs because as the interval time increases, fewer packets are generated, which reduces the network load.

5 Conclusion

In this chapter, a three-layer Micro-ANP protocol architecture for UANs is introduced. Furtherly, a kind of digital fountain code which is called as RLT is presented. RLT is characterized by small degree distribution and recursive encoding, so RLT reduces the complexity of encoding and decoding. Based on the Micro-ANP architecture and RLT code, a handshake-free reliable transmission mechanism-RCHF is presented. In RCHF protocol, frames are forwarded according to the state of the receiver which can avoid the sending-receiving collisions and overhearing collisions. Simulations show that RCHF protocol can provide higher delivery ratio, throughput and lower end-to-end delay.

As a new trends, how to combine the specific underwater application scenarios, transform the negative factors of UANs into favorable factors is an interesting research. For example, the mobility of nodes brings about extra routing overhead, and reduces end-to-end performance. However, the mobility of Autonomous Underwater Vehicle (AUV) and the policy of cache-carry-forward help to improve the data forwarding rate.

Meanwhile, under the precondition of less resource consumption, guaranteed channel utilization and network throughput, combining the technologies of channel coding, cognitive underwater acoustic communication, data compression and post quantum public key cryptography, studying on secure and reliable data transmission is another future work.

Acknowledgments

This work is supported by the National Natural Science Foundation Projects of China (61162003), Key Laboratory of IoT of Qinghai Province, Qinghai Office of Science and Technology (2015-ZJ-904), Hebei Engineering Technology Research Center for IOT Data acquisition & Processing.

References

[1] Zhong Zhou, Zheng Peng, Jun-Hong Cui, *et al.* Handling Triple Hidden Terminal Problems for Multi-Channel MAC in Long-Delay Underwater Sensor Networks[C]//Proceedings of International Conference on Computer Communications (INFOCOM), San Diego, USA, IEEE Computer Society Pr. 2010:1-21.

[2] Dario Pompili, Ian F. Akyildiz. A Multimedia Cross-Layer Protocol for Underwater Acoustic Sensor Networks[J]. IEEE Transaction on Wireless Communications, 2010, 9(9): 2924-2933.

[3] Dario Pompili, Tommaso Melodia, Ian F. Akyildiz, Distributed Routing Algorithms for Underwater Acoustic Sensor Networks[J]. IEEE Transaction on Wireless Communications, 2010, 9(9):2934-2944.

[4] Chenn-Jung Huang, Yu-Wu Wang, Hsiu-Hui Liao. A Power-Efficient Routing Protocol for Underwater Wireless Sensor Networks[J]. Applied Soft Computing, 2011, 11(2): 2348-2355.

[5] Zhong Zhou, Jun-Hong Cui. Energy Efficient Multi-Path Communication for Time-Critical Applications in Underwater Sensor Networks[C]// Proceedings of the 9th ACM International

Symposium on Mobile Ad Hoc Networking and Computing, Hong Kong, China, ACM New York, USA, 2008:1-31.

[6] K. Hao, Z. Jin, H. Shen, Y. Wang. An Efficient and Reliable Geographic Routing Protocol Based on Partial Network Coding for Underwater Sensor Networks. Sensors, 2015, 15, 12720-12735.

[7] X. Du, K. Huang, S. Lan. LB-AGR: Level-Based Adaptive Geo-routing for Underwater Sensor Networks. The Journal of China Universities of Posts and Telecommunications, 2014, 21(1):54-59.

[8] X. Du, C. Peng, X. Liu, Y. Liu. Hierarchical Code Assignment Algorithm and State-based CDMA Protocol for UWSN. China Communications, 2015, 12(3): 50-61.

[9] X. Du, K. Li, X. Liu, Y. Su. RLT code based handshake-free reliable MAC protocol for under-water sensor networks. Journal of Sensors, Volume 2016, http://dx.doi.org/10.1155/2016/3184642.

[10]X. Du, X. Liu, Y. Su. Underwater Acoustic Networks Testbed for Ecological Monitoring of Qinghai Lake[C]//Proceedings of oceans16 Shanghai, 2016.4.10.

[11]Y. Dong, P. Liu. Security Consideration of Underwater Acoustic Networks. [C]//Proceedings of International Congress on Acoustics, ICA 2010.

[12]Y. Cong, G. Yang, Z. Wei, W. Zhou. Security in Underwater Sensor Network. [C]//Proceedings of International Conference on Communication and Mobile Computing, 2010, 1(1):162-168.

[13]G. Dini, A. Lo Duca. A Cryptographic Suite for Underwater Cooperative Applications[C]//Proceedings of IEEE Symposium on Computers & Communications, 2011:870-875.

[14]C. Peng, X. Du, K. Li, M. Li. An Ultra Lightweight Encryption Scheme in Underwater Acoustic Networks, Journal of Sensors, Volume 2016, http://dx.doi.org/10.1155/2016/8763528.

[15] X. Du. Micro-ANP protocol architecture for UWSN. China Patent ZL201210053141.0 Nov. 2014.

[16] Marc¸al Molins, Milica Stojanovic. Slotted FAMA: a MAC Protocol for Underwater Acoustic Networks[C]//Proceedings of IEEE OCEANS'06, Singapore, 2006:16-22.

[17]I. REED and G. Solomon. Polynomial Codes over certain finite fields. Journal of the Society for Industrial and Applied Mathematics, June 1960, 300-304.

[18]M. Luby, M. Mitzenmacher, A. Shokrollahi, D. Spielman. Practical Loss-Resilient Codes. In ACM STOC, 1997:150-159.

[19]P. Xie, Z. Zhou, Z. Peng, J. Cui, Z. Shi. SDRT: A reliable data transport protocol for underwater sensor networks. Ad Hoc Networks ,2010, 8(7):708-722.

[20]H. Mo, Z. Peng, Z. Zhou, M. Zuba, Z. Jiang, J. Cui. Coding based Multi-hop Coordinated Reliable Data Transfer for Underwater Acoustic Networks: Design, Implementation and Tests[C]// Proceedings of Globecom 2013, Wireless Network Symposium: 5066-5071

[21]D. J. C. MacKay. Fountain Codes[C]//Proceedings of IEEE Communications, 2005, 152,(6): 1062-1068.

[22]A. Shokrollahi. Raptor codes. Information Theory. IEEE Transactions, 2006, 52(6):2551-2567.

[23]M. Luby. LT codes[C]//Proceedings of the 43rd Annual IEEE Symposium on Foundations of Computer Science, 2002, 12: 271-280.

[24]P. Xie, J.-H. Cui, L. Lao. VBF: Vector-based forwarding protocol for underwater sensor networks[C]//Proceedings of IFIP Networking May 2006.

[25]Yan H, Shi Z.J, Cui J H. DBR: depth-based routing for underwater sensor networks[C]//Proceedings of IFIP Networking, Singapore, 2008:72-86.

[26]Niculescu D, Nath B. DV based positioning in ad hoc networks. Journal of Telecommunication Systems, 2003, 22(1/4): 267- 280.