International Journal of Advances in Engineering and Management (IJAEM)
Volume 5, Issue 9 Sep 2023, pp: 337-343 www.ijaem.net ISSN: 2395-5252

Performance Analysis of Packet Transmission and Timeout Ratio Using Ml Techniques

V. Gokul, Dr. M. Shanmugapriya

Research Scholar, Department of Computer Science, Park's College (Autonomous), Tirupur, Tamil Nadu, India
Assistant Professor, Department of Computer Science (UG), Kongu Arts and Science College (Autonomous), Erode, Tamil Nadu, India

---

Date of Submission: 15-09-2023 Date of Acceptance: 25-09-2023

ABSTRACT:
Classic Karn's method is used in a network to estimate packet transmission round-trip time between sender and recipient. This is learned by packet timeout analysis. Karn's technique modifies the timeout depending on the average round-trip time of successful packets. The Karn approach may reduce undesired retransmissions and improve network performance by changing the timeout setting. Karn's method's main drawback is that it estimates RTT and timeout using only successful packets. The fundamental drawback of Karn's method. It does not count lost or dropped packets until they are retransmitted. Since the timeout has passed, the sender will not get an acknowledgement for a lost packet. Since the Karn algorithm does not obtain an ACK for the lost packet, it does not adjust the timeout number using its RTT. This recommended approach uses cross-layer Dynamic Adaptive ARQ. The network and receiver input determine ARQ settings like timeout and retransmissions, which this method modifies dynamically. For each packet transfer, the Dynamic Adaptive ARQ algorithm predicts the best ARQ settings using machine learning. Past network conditions and packet loss statistics inform these projections. DA-ARQ adjusts ARQ parameters like the retransmission timeout and number of retransmissions based on network and receiver information. DA-ARQ also adjusts transmission redundancy based on network error rates.

I. INTRODUCTION

The term Automatic Repeat Request, which is also often referred to as Automatic Repeat Query, is referred to as ARQ in its shortened form. An error-control method known as ARQ is used in communication systems that are capable of functioning in both directions simultaneously. It is a set of error-control methods that are used to ensure reliable data transmission over a source or service that is not trustworthy. The phrase "error control techniques" refers to this collection. Within the Open Systems Interconnection architecture, these protocols are located at both the Transport Layer and the Data Link Layer. These protocols are responsible for the automatic retransmission of any packets that are found to be corrupted or missing while the transmission is in progress, and they assume this obligation during transmission. The basic goal of these protocols is to guarantee that the sender will obtain an acknowledgement from the receiving end before the timeout threshold is reached, signaling that the document or packet was properly received. A timeout is a fixed period of time, defined in advance, during which the receiver is obligated to deliver the acknowledgment to the sender. In the case that a timeout occurs, which indicates that the receiver does not get the acknowledgement within the authorized amount of time, it is considered that either the timer or the packet was corrupted or lost while it was being delivered. This is because the timeout indicates that the recipient did not receive the acknowledgment within the given amount of time. As a consequence of this, the sender will resend the packet, and the protocols will ensure that the operation described above will continue to be carried out until the correct packet is sent. ARQ protocols are useful in a wide range of settings because they are able to maintain reliable transmissions even when the upstream sources, they are connected to are unstable. Shortwave radio is where these protocols find their primary use, since it is the only medium that can reliably ensure the transfer of signals.

1.1 Types of ARQ

Stop And Wait ARQ:

ARQ is an acronym that stands for "stop and wait," and it is a method that is used in two-

International Journal of Advances in Engineering and Management (IJAEM)
Volume 5, Issue 9 Sep 2023, pp: 337-343 www.ijaem.net ISSN: 2395-5252

way communication systems to transfer information between two coupled devices (a sender and a receiver). This method is also referred to as the alternating technique in certain circles. It is also known as a stop-and-wait ARQ due to the fact that the aim of this form of protocol is to send a single picture at a time, and because of this, another name for it is single-image ARQ. Once a frame or packet has been sent, the sender will not send any more packets until it has received an acknowledgement from the receiving device. This means that the sender will wait to send any additional frames or packets. In addition, the sender will save a copy of the packet once it has been successfully sent. The receiver is responsible for sending an acknowledgement when it has successfully obtained the frame that was requested. If the sender does not get an acknowledgement within the predetermined length of time, which is also referred to as the timeouts, then the sender will resend the same packet. The timer starts again from scratch after the transmission of each frame has been finished. This kind of control is known as a stop-and-wait ARQ because it is used in situations such to the one that was just described, which is an example of a stop-and-wait scenario.

Go Back-N ARQ:

The ARQ protocol is a kind of protocol in which the transmitting process continues to send several frames or transmissions even after it has not received an acknowledgement message from the receiver. This is done in order to increase the likelihood that the data will be successfully delivered. This is carried out in a manner that is consistent with the ARQ protocol. The sequence number that corresponds to the next packet that the receiver process anticipates receiving is included in the acknowledgement that it sends to the sender. This number is sent by the receiver process. The receiver process is responsible for maintaining a record of this sequence number. Any data packet that does not include the expected sequence number will be thrown away by the receiver. Instead, the receiver will provide an acknowledgement for the most recent frame that was successfully received in the right format. In point of fact, there are only two circumstances in which a frame's sequence number won't match: either it is a copy of an already existing frame or it is an out-of-order message that has to be sent later. In any instance, the person who is receiving it is aware of this condition and responds with an acknowledgement signal that is appropriate for the situation. After the sender has completed sending all of the frames that are included within its window, it will identify which frames have been lost since the first lost frame and which frames have been gained after that point. After then, it will start the process all over again by rewinding to the sequence number of the most recent acknowledgement signal that it received from its receiving pr and then moving forward from there. This sort of system will re-transmit the whole send window's following frames as well as the frame that was determined to have been corrupted or lost in the event that a frame is found to have been damaged or lost. This is done in the event that a frame is found to have been damaged or lost. This is the one and only drawback associated with using a system like this. This protocol is much more effective than the Stop and wait ARQ approach due to the absence of any waiting time that is required during its execution.

Selective Repeat ARQ/Selective Reject ARQ:

Even after it is determined that a frame has been damaged or lost, the transmitting process will continue since it is governed by the Selective Repeat ARQ/Selective Reject ARQ protocol mechanism. In contrast to this, the Go-Back-N protocol mechanism, which halts the transmitting process whenever it determines that a frame has been damaged or lost, continues until the problem is resolved. This is done by the following method: the receiver process keeps a note of the number of sequences of the earliest frame that it has missed receiving and sends the correct sequence number along with the acknowledgement signal. In this manner, the goal is successfully achieved. In the case that a whole frame does not arrive at the recipient's end, the sender will continue to send successive frames until it has used up all of the remaining space in its window. This will occur even if the sender detects that a full frame did not arrive at the recipient's end. Once this step of mistake repair has been finished, the operation will continue from the place where it was halted because of the step to fix the error. In contrast to the Go back-N protocol, this one does not repeat the sending of a packet after it has already been sent.

1.2 Advantages of ARQ

- Error-detection and repair procedures are fairly basic when compared to other approaches.
- Decoding equipment may be made much more straightforward when compared to other systems.

International Journal of Advances in Engineering and Management (IJAEM)
Volume 5, Issue 9 Sep 2023, pp: 337-343 www.ijaem.net ISSN: 2395-5252

1.3 Disadvantages of ARQ

- Since there is a chance that information will be lost due to the high error rate in the channel, the system's efficiency or productivity will suffer.
- If a media or a channel has a high error rate, this could lead to an excessive amount of information being sent in frames or packets.

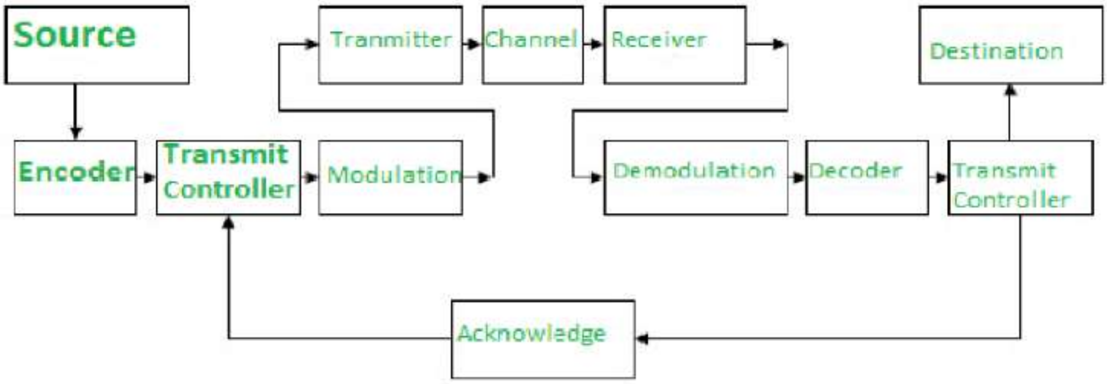

Fig.1.1 ARQ Process

II. RELATED WORK

Recent advancements in the field of wireless networking have made it much easier to provide trustworthy solutions for a wide range of applications across many different domains [1, 2]. These solutions can be used in anything from large-scale sensor networks to infrastructure monitoring systems to roadside networks for automotive applications. Without the requirement for a fixed infrastructure, nodes in a mobile Ad-Hoc network may establish active and visible connections with one another. Every interaction node in this network may act as either a transceiver or a relay thanks to the multi-hop transmission paradigm. To guarantee theoretical communication between any possible pair of sources and targets, this is required. Ad-hoc wireless networks provide unique challenges when it comes to transmission scheduling due to a variety of factors, one of which is the need to be able to quickly adjust to new circumstances. Several researchers have been working on this problem [3] as of late, with the main goal of optimizing radio use. Primarily, it involves optimizing the throughput of serial transmissions by using a schedule of minimal duration that meets a specified class of traffic needs in the smallest possible number of time slots within a nominally periodical time frame [4]. To achieve this, a timetable of minimal duration is used. Each time slot has its own separate, non-overlapping group of broadcasts, or transmissions, based to the so-called actual interference concept. If the signal-to-interference-plus-noise ratio (SINR) at the desired receivers is greater than a predetermined value, then the broadcast was successful.

For each connection, several packets must be scheduled using the traditional minimum-length scheduling approach [3]. Suitable sets may be added to the frame in any sequence, and the outcome will be the same. However, the order in which packets are transmitted might affect their total delay. Due to the method's reliance on the assumption of traffic periodicity, repeating the frame guarantees that all link transfers planned inside the window are used to their full potential. From a latency perspective, this approach is no longer viable if preserving frequency in the transfer of a batch of packets is a priority. Two approaches are presented in [5] with the intention of achieving this by speeding up the delivery of packets with many source-destination pairs. The first method employs a standard packet-forwarding paradigm, where appropriate parameters are used for every window of time inside a frame. The second method integrates forward interference cancellation with cooperative packet forwarding. Multiple transmitters may use CPF to emit the same packet simultaneously and a receiver can combine the resulting signals to improve the SINR (signal-to-interference-plus-noise ratio). After delaying and confirming the successful reception of a packet, a node may utilize FIC to eliminate the interference generated by the packet's transmission. However, there are a variety of factors that affect the efficiency of communication scheduling including packet routing in a real-world wireless Ad-Hoc network. The deployment environment imposes constraints on a number of elements, including propagation conditions, environmental characteristics, and node locations [7]. As a consequence of the topographical profile characteristics that prevent full paths from being formed for these packets, it is not assured that the optimizing of the speed of delivery of packages that are sent into the network will be achieved using the two indicated approaches. This is because the properties of the terrain profile mentioned above make it impossible to build complete routes.

The network's performance may be enhanced by striking a balance between the previously stated environmental components and constraints [8, 9] in order to guarantee the fastest possible end-to-end delivery of a certain number of inserted packets. This is important because there may be times when you have competing objectives. Therefore, the most popular approach to enhancing this network's functionality is to determine the optimal node locations. However, this challenge is made far more challenging in 3D systems due to the fact that its resolution is context-dependent. The propagation loss due to flaws in the terrain profile over a three-dimensional environment also has to be accounted for, therefore a thorough analysis is necessary. Based on these results, this research aims to provide a strategy for improving transmission scheduling in practical wireless Ad-Hoc networks by using a multi-hop broadcast model. To reach our objective, we must address two related problems. The first goal is to increase the throughput at which insertion packets are delivered, and the second goal is to decrease the total round-trip time for a set of packets. The SINR interference model is also considered with the goal of optimizing the outbound transmission schedule. The packet delivery rate is defined as the ratio of the total number of packets sent to the total number of sequences that were delivered successfully. However, the overall amount of time it takes to send a group of packets down their individual paths is known as the end-to-end latency.

III. PROPOSED METHODOLOGY

3.1 KARN'S Algorithm

Karn's method eliminates the difficulty of calculating precise message round-trip times using the Transmission Control Protocol, a protocol used in networked computers. The method, termed as the Karn-Partridge algorithm, was proposed in 1987 by Phil Karn and Craig Partridge. TCP round-trip estimates may be imprecise due to the uncertainty caused by retransmitted segments. To determine the round-trip time, we subtract the time a segment was sent from the time it received an acknowledgment. However, when packets are re-transmitted, this calculation becomes ambiguous because the acknowledgment could be in response to either the initial transmission of the part or the subsequent re-transmission. Karn's Algorithm doesn't take retransmitted segments into account when calculating the round-trip time. The estimation of round-trip time is based only on unmistakable acknowledgments, which are confirmations of single-delivery of segments.

This may potentially be a problem since Karn's approach has been simplified too much in this implementation. Think about what happens after a considerable latency increase when TCP transmits a segment. Whenever the round-trip time estimate for a segment exceeds a certain threshold, TCP issues a delay and retransmits the segment. TCP will continue to retransmit every segment without compensating for the additional delay if it does not take into account the round-trip duration of all retransmitted packets. TCP performance is enhanced under less-than-ideal network conditions thanks to Karn's Algorithm. By monitoring the rates of change in RTOs and RTTs, we can quickly locate the source of network issues and fix them. If that happens, maybe the apps we support will run more smoothly and with less reliance on Karn's Algorithm. There is an issue when a packet has to be resent. The RTO is reset if a timeout occurs while the packet is being resent. Once the message is resent with the increased RTO, an acknowledgement is received. The acknowledgement that is received is the same whether it pertains to the first or second transmission. Retransmission ambiguity describes the issue at hand.

The retransmission ambiguity issue occurs when a sender retransmits data and gets an acknowledgement for it. The sender does not know whether the ACK they got is for the new packet or the one they reprinted because of a previous loss. If the receiving end acknowledges the sending end, the RTT sample will be significantly longer above the current SRTT value. It is probable that the original payload or ACK was lost if the receiver confirms the retransmitted packet with an RTT that is lower than the current SRTT value. When there is resend ambiguity, RTT values are ignored rather than included into SRTT per the first portion of Karn's Algorithm. TCP gives each RTO a "backoff factor" for each retransmission, doubling the elapsed time between retries each time. Once a data transfer has been completed successfully without the need for retransmission, only then will the backoff factor be reset. Together, these methods make up the second half of Karn's Algorithm. By

International Journal of Advances in Engineering and Management (IJAEM)
Volume 5, Issue 9 Sep 2023, pp: 337-343 www.ijaem.net ISSN: 2395-5252

following these steps, the network is protected against an influx of unnecessary duplicate packets, speeding up its recovery from congestion issues. It also prevents the loss of any critical RTT data. The initial part of Karn's Algorithm disregards all RTT values within the RTO backoff due to the fact that data is retransmitted. It is possible to include the RTT measurement, which is expected to be close to the prevailing RTO value when data is successfully delivered without retransmission, into SRTT.

3.2 DA-ARQ Algorithm

The traditional implementation of Karn's Algorithm gets rid of receiver information like as ACKs and NACKs, which might provide vital details on the status of the network at any given moment. Changing the retransmission rate depending on the conditions of the network is one of the advanced ways that may be used to improve reliability, reduce latency, and boost efficiency. One such strategy is dynamic adaptive ARQ. The classic Karn's Algorithm was an excellent method back in the day, but it is no longer viable in the complex and ever-changing network environments of today. The complexity of modern networks necessitates the use of more up-to-date, versatile, and flexible solutions, such as DA-ARQ, which are better equipped to deal with this complexity and may boost network performance. Because the traditional Karn's Algorithm does not modify its timeout value in response to shifting conditions in the network environment, it is possible that ineffective retransmissions and increased delay will occur as a consequence. Because it relies on a predetermined timeout number that is determined by the network's anticipated round-trip time, the classic Karn's Algorithm is regarded to be out of date when it comes to calculating packet timeout analysis in modern networks. This is one of the reasons why the algorithm is no longer used. Examining how well a method deals with packet timeouts and retransmissions is one of the tasks involved in determining how well the DA-ARQ method handles the evaluation of packet transfer delivery timeouts that occurs during the transmission of packets using that method. Assess the effectiveness of the DA-ARQ strategy in dealing with packet timeouts during transmissions by analyzing the data that was obtained and analyzing it. Calculations need to be made about metrics such as the packet delivery ratio, the average delivery time, and the retransmission count. DA-ARQ, which stands for "Dynamic Adaptive Automatic Repeat Request," is a method of network communications that increases the dependability of data transmission via channels that are not encrypted. This combines the ARQ protocol, which stands for Automatic Repeat request, with various ways of dynamic adaptability.

Within DA-ARQ, dynamic adaptation techniques are used in order to modify the approach to retransmission dependent on the conditions of the network. Continuously evaluating channel quality, latency, and congestion allows it to make educated conclusions about the state of the network. The retransmission method used by DA-ARQ may be adjusted depending on the conditions of the network. It may change the amount of retransmissions attempted, tweak the timeout intervals, or use forward error correction (FEC) procedures in order to recover from mistakes. DA-ARQ makes use of dynamic adaptation methodologies in order to modify the retransmission strategy in accordance with the circumstances of the network and the feedback at the packet level. In order to arrive at sound conclusions, it keeps a close eye on factors such as the rate of packet loss, the latency, and the congestion. The retransmission method used by DA-ARQ is dynamically adjusted on the packet level based on the conditions of the network as well as the replies received from the receiver. In order to recover from errors, it could change the number of times it attempts to retransmit individual packets, vary the timeout intervals for each packet, or use forward error correction methods on certain packets. In addition to this, DA-ARQ takes into consideration the congestion experienced on the network at the packet level. In the event that congestion is detected, the retransmission rate for certain packets may be lowered, or congestion management strategies may be used to lessen the impact of the existing congestion and prevent it from spreading. Instead of resending the whole data stream, DA-ARQ works on a more compact scale, which enables the selective retransmission of certain packets that have been identified as dropping or erroneous rather than sending the entire data stream again. At the level of the packet, the retransmission operation is continued until either all of the packets are successfully delivered to the receiver or a predetermined maximum number of retransmissions for specific packets has been achieved. DA-ARQ alters the retransmission method in accordance with the observable conditions of the network and the feedback received at the packet level. It may vary the amount of retransmission attempts for individual packets, modify the timeout periods for each packet, or employ forward error correction (FEC) techniques on specific packets in order to recover from failures. DA-ARQ, which stands for Distributed Adaptive Automatic Repeat Request, is a paradigm

for wireless data transmission that improves efficiency by minimizing the occurrence of errors and the need for retransmissions.
During the process of data transmission, DA-ARQ employs error detection strategies, such as CRC (Cyclic Redundancy Check), to identify whether or not any mistakes have occurred. The retransmission method used by DA-ARQ is contingent upon the characteristics of the channel. It does this by dynamically adjusting the amount of retransmissions in accordance with the quality of the wireless channel as it is seen. In FEC, redundant information is added to the data that is being sent in order to assist error detection and repair at the receiver end. These approaches may also be used in conjunction with DA-ARQ. It also helps to lessen the need for retransmissions, which is a useful benefit.The DA-ARQ method augments the original data with redundant data, which enables packet recovery without the need of a retransmission request being made. By resending just the packets that were corrupted rather than the complete sequence, it may be possible to achieve a greater degree of dependability and data integrity than would otherwise be possible. By using more advanced methods like as forward error correction (FEC) and interleaving, it is able to deal with more complex scenarios, such as the duplication and reordering of packets. These methods also modify the settings in reaction to altering network circumstances, such as varying error rates and use of available bandwidth. The dynamic modification of the transmission rate and modulation mechanism is the primary means by which it enhances resource use. Because of this, the system's efficiency increases, and the amount of energy that it consumes drops. The DA-ARQ method is more successful than classic Karn's and other algorithms since it makes use of a dynamic strategy to modify the retransmission rate according to the conditions of the network. This results in a decreased need for retransmissions, which in turn contributes to an overall improvement in the system's efficiency. By altering the retransmission rate according to the conditions of the network, the DA-ARQ approach is able to significantly cut down on the latency. This results in an increase in the system's overall efficiency as well as a reduction in the amount of time needed to convey the data. The DA-ARQ method is more dependable than the traditional Karn's approach since it makes use of a feedback mechanism to identify and address any errors that may have been made. Because of this, the number of mistakes that occur is decreased, and the system's overall dependability is improved. The DA-ARQ strategy can efficiently handle a greater number of users and adapt to ever-shifting network circumstances, making it the more effective strategy.

IV. CONCLUSION

DA-ARQ uses a dynamic approach to adjust retransmission rates depending on network circumstances, outperforming classic algorithms like Karn's. This reduces retransmissions and improves system efficiency. The DA-ARQ method reduces latency by adapting the retransmission rate to network conditions. This boosts system performance and speeds data transmission. The DA-ARQ technique is more reliable than Karn's since it uses feedback to find and rectify faults. This reduces mistakes and enhances system reliability. DA-ARQ is more scalable than Karn's algorithm since it can manage more users and adapt to network changes. Large-scale networks may use it. The DA-ARQ technique is more secure since it encrypts and decrypts data during transmission. This safeguards data privacy and prevents illegal access. DA-ARQ is a reliable packet timeout analysis method that reduces latency, improves reliability, and improves retransmissions. Further it can be extended in terms of providing consistent transmission without overhead.

REFERENCES

[1]. Antonio Capone, Yuan Li, MichałPióro, Di Yuan, Minimizing end-to-end delay in multi-hop wireless networks with optimized transmission scheduling, Ad Hoc Networks, Volume 89, 2019, Pages 236-248.

[2]. Md. Forkan Uddin, Energy efficiency maximization by joint transmission scheduling and resource allocation in downlink NOMA cellular networks, Computer Networks, Volume 159, 2019, Pages 37-50

[3]. Arun, V., Alizadeh, M., Balakrishnan, H.: Starvation in end-to-end congestion control. In: Proceedings of the ACM SIGCOMM 2022 Conference, pp. 177–192 (2022)

[4]. Arun, V., Arashloo, M.T., Saeed, A., Alizadeh, M., Balakrishnan, H.: Toward formally verifying congestion control behavior. In: SIGCOMM 2021 (2021)

[5]. Bensley, S., Thaler, D., Balasubramanian, P., Eggert, L., Judd, G.: Data Center TCP (DCTCP): TCP congestion control for data centers, October 2017. Accessed 15 Mar 2023

[6]. Aboba, B., Wood, J.: Authentication, authorization and accounting (AAA) transport profile, June 2003. Accessed 21 Mar 2023
[7]. Alqudah, Nour&Yaseen, Qussai. (2020). Machine Learning for Traffic Analysis: A Review. Procedia Computer Science. 170. 911-916. 10.1016/j.procs.2020.03.111.
[8]. Aldhyani, Theyazn. (2020). A review of network traffic analysis and prediction techniques.
[9]. S. Fu, Y. Zhang, Y. Jiang, C. Hu, C.-Y. Shih, and P. Marron, "Experimental study for multi-layer parameter configuration of WSN links," in IEEE ICDCS, 2015.
[10]. S. Fu, Y. Zhang, M. Ceriotti, Yuming Jiang, M. Packeiser, and P. Marron, "Modeling packet loss rate of IEEE 802.15.4 links in diverse environmental conditions", IEEE WCNC, 2018
[11]. Y. Zhang, S. Fu, Y. Jiang, M. Ceriotti, M. Packeiser, and P. Marron, "An LQI-Based Packet Loss Rate Model for IEEE 802.15.4 Links", IEEE PIMRC, 2018
[12]. T. Wiki., "Cc2420 hardware and software acks," http://http://tinyos. stanford.edu/tinyos-wiki/index.php/PacketLink.
[13]. S. Fu and Y. Zhang, "CRAWDAD data set due/packet-delivery (v. 2015- 03-30)," Downloaded from http://crawdad.org/due/packet-delivery/, Mar. 2015.
[14]. Cheng, Y., Cardwell, N., Dukkipati, N., Jha, P.: The RACK-TLP loss detection algorithm for TCP, February 2021
[15]. Hurtig, P., Brunstrom, A., Petlund, A., Welzl, M.: TCP and Stream Control Transmission Protocol (SCTP) RTO restart, February 2016