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Hybrid Congestion Control Mechanisms for Next-Generation Communication Networks

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***Abstract:** In the rapidly evolving realm of next-generation communication systems, characterized by ultra-low latency and high-speed data transmission, effectively managing network congestion remains a critical challenge. Traditional congestion control mechanisms often struggle to meet the demands of these advanced environments. This paper presents a novel approach that integrates both delay and loss metrics, specifically designed for 5G and beyond. By utilizing real-time variations in delay and packet loss as indicators of congestion, the proposed method enables dynamic adjustments of data transmission rates to mitigate congestion proactively. This strategy aims to minimize packet loss, reduce latency, and enhance throughput, addressing the needs of modern applications such as IoT and autonomous vehicles. Extensive simulations demonstrate significant improvements in network efficiency and reliability compared to traditional algorithms, contributing to the development of adaptive congestion control mechanisms that ensure consistent, high-quality service in complex network conditions.*

***Keywords:** Congestion Control, Delay-Based Algorithm, Packet Loss, TCP Variants, Network Performance.*

I. INTRODUCTION

The advent of next-generation communication systems, including 5G and emerging 6G networks, marks a transformative shift in the way data is transmitted and received across the globe. These advanced networks promise unprecedented capabilities such as ultra-high data rates, near-zero latency, and the ability to support billions of connected devices simultaneously [3]. Such innovations are set to revolutionize industries ranging from healthcare and transportation to entertainment and smart cities, enabling applications that were once considered science fiction. However, these advancements also bring about significant challenges, particularly in the realm of network congestion control.

Traditional congestion control mechanisms [1], which have been effective in earlier generations of networks, often fall short in the face of the complex and demanding requirements of next-generation systems. The sheer volume of data, coupled with the need for real-time communication and the diverse range of applications, places immense strain on network resources. Maintaining low latency and minimizing packet loss are critical for ensuring a seamless user experience in applications such as augmented reality, autonomous vehicles, and the Internet of Things (IoT) [12]. One of the key limitations of conventional congestion control approaches is their reliance on packet loss as the primary indicator of network congestion. While effective in many scenarios, this reactive approach can lead to suboptimal performance in next-generation networks, where the speed and scale of data transmission necessitate more proactive and adaptive strategies. As a result, there is a growing need for congestion control mechanisms that can better align with the unique characteristics of these advanced communication systems.

In this context, delay-based congestion control emerges as a promising alternative [4]. By using variations in network delay as an early indicator of congestion, this approach allows for the dynamic adjustment of data transmission rates, helping to prevent congestion before it leads to significant packet loss or increased latency. This proactive strategy is particularly well-suited to the demands of next-generation networks, where maintaining high throughput and low latency is crucial.

This paper proposes a novel delay-based congestion control approach specifically designed for next-generation communication systems. The proposed method leverages real-time delay measurements to optimize network performance, offering a more responsive and adaptive solution compared to traditional loss-based mechanisms. Through extensive simulations, this study demonstrates the potential of delay-based congestion control to enhance network efficiency, reliability, and overall quality of service in the context of next-generation communication networks. The findings of this research contribute to the ongoing development of more sophisticated and effective congestion control strategies, ultimately supporting the successful deployment and operation of 5G and beyond.

A. Highlights of the Proposed Delay-Based Congestion Control Solution

- **Proactive Congestion Detection:** Unlike traditional loss-based approaches that react after congestion occurs, this solution uses real-time delay measurements as an early indicator of network congestion, allowing for dynamic adjustments to transmission rates before packet loss and significant latency increases occur.
- **Adaptability to Next-Generation Networks:** The proposed

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method is specifically designed for the high-speed, low-latency demands of 5G and emerging 6G networks, where the conventional congestion control strategies may struggle to maintain performance due to the complexity and scale of these systems.

- **Low Latency and High Throughput Maintenance:** By reacting to variations in delay, the algorithm maintains optimal throughput while keeping latency minimal, making it ideal for latency-sensitive applications like augmented reality (AR), virtual reality (VR), autonomous vehicles, and IoT.
- **Reduced Packet Loss:** With delay-based adjustments to transmission rates, the system prevents congestion build-up, resulting in reduced packet loss compared to loss-based control mechanisms.
- **Real-Time Performance Optimization:** The algorithm continuously monitors network conditions and dynamically adjusts based on real-time delay feedback, ensuring high network efficiency under varying traffic loads and conditions.

The organization of this paper is structured as follows: The introduction section provides an overview of the challenges in next-generation communication systems, focusing on congestion control and introducing the proposed delay-based approach. The Related Work section reviews existing congestion control mechanisms, including traditional and delay-based strategies, highlighting their relevance to advanced networks. The Proposed Delay-Based Congestion Control Approach section details the design and operation of the algorithm, explaining how real-time delay variations are used for proactive congestion management. The Simulation Setup and Results Analysis section describes the simulation environment and the metrics used for evaluation. The section summarizes the key findings and outlines potential directions for further research and development in this area.

II. RELATED WORK

The evolution of congestion control mechanisms has been pivotal in adapting to the rapidly changing demands of communication networks. As networks have transitioned from early TCP/IP-based architectures to the highly complex and diverse environments seen in next-generation systems, the limitations of traditional congestion control approaches have become increasingly apparent. In particular, the reliance on packet loss as the primary congestion signal in conventional TCP congestion control algorithms, such as TCP Reno [1] and TCP New Reno [2], often leads to inefficient performance in high-bandwidth, low-latency environments typical of 5G and emerging 6G networks.

Delay-based congestion control has emerged as a promising alternative, addressing the shortcomings of loss-based methods by utilizing round-trip time (RTT) variations as early indicators of network congestion. TCP Vegas [3], one of the earliest delay-based algorithms, introduced the concept of using RTT to adjust the sending rate proactively, thereby preventing congestion before it results in significant packet loss. While TCP Vegas demonstrated improved performance in certain scenarios, it was also found to be overly conservative in high-speed networks, limiting its throughput potential.

Subsequent efforts to refine delay-based congestion control have led to the development of more sophisticated algorithms, such as TCP Veno [4], Compound TCP [5] and TCP Illinois [6]. TCP Veno, for example, combines delay and loss signals to enhance performance in wireless networks, where packet loss can occur independently of congestion due to the unreliable nature of the medium. TCP Illinois [6] integrates both delay and loss metrics to dynamically adjust the congestion window, achieving a balance between high throughput and low latency, particularly in high-speed, long-distance networks. Compound TCP [5], adopted by Microsoft, merges delay-based and loss-based approaches, optimizing performance across a wide range of network conditions.

Recent advancements in congestion control research have led to the development of algorithms tailored specifically for the unique challenges posed by next-generation communication systems. BBR [7] (Bottleneck Bandwidth and Round-trip propagation time) introduced a novel approach by estimating the available bandwidth and RTT independently of packet loss, offering significant improvements in throughput and latency for modern high-speed networks. BBRv2 [8] builds on this by addressing fairness issues and incorporating explicit congestion notification (ECN) to improve performance in loss-prone environments.

In the context of 5G and beyond, several new approaches have emerged. For instance, TCP Recent Acknowledgment and TCP Prague are recent innovations designed to improve performance in high-speed, low-latency networks. TCP RACK [9] introduces a more efficient loss recovery mechanism by using recent ACKs to detect lost packets more quickly, thus reducing recovery times and improving overall throughput. TCP Prague [10], part of the Low Latency Low Loss Scalable Throughput (L4S) initiative, aims to provide ultra-low latency for interactive applications by leveraging Explicit Congestion Notification (ECN) and a scalable congestion control mechanism.

Moreover, machine learning-based congestion control algorithms are gaining traction, offering adaptive and intelligent approaches to manage network congestion. Congestion Control using Deep Reinforcement Learning [11] (CoDel-DRL), for example, has shown promise in dynamically optimizing congestion control parameters based on real-time network conditions, resulting in improved QoS and resource utilization in next-generation networks.

Verma et al. [12] address the growing issue of network congestion caused by the exponential increase in connected devices within the IoT. Recognizing that the TCP is essential for reliable data transmission in both wired and wireless networks, they highlight the protocol's challenges in dynamically adjusting transmission rates in response to abrupt changes in network conditions, such as delays and bandwidth fluctuations. Mishra et al. [13] provides a comprehensive survey of congestion control algorithms essential for managing the increasing network congestion in the IoT. It discusses the limitations of current TCP versions in handling diverse IoT devices and emphasizes the need for advanced protocol stacks to ensure effective

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communication. Verma et al. [14] introduce a new TCP variant called Delay-based Adaptive Congestion Control (DACC) aimed at improving congestion management in networks. DACC addresses these issues by factoring in background flows and quickly adapting to current network conditions, allowing for more efficient transmission rate adjustments. Simulation results demonstrate that DACC significantly enhances goodput, reduces Packet Loss Ratio (PLR), and improves both inter- and intra-protocol fairness, ultimately facilitating faster file transfers. N. Mishra et al. [15] conduct an in-depth comparison of various congestion control algorithms utilized in the transport layer, focusing on their performance metrics and effectiveness in managing network congestion [16]. The authors analyze key algorithms, including traditional TCP variations, examining their mechanisms for controlling the CWND, RTT, and RTO [17]. By evaluating the strengths and weaknesses of each algorithm in different network scenarios, the paper aims to provide insights into their applicability and efficiency for various applications [18].

The integration of delay-based congestion control with emerging technologies such as software-defined networking (SDN) and network function virtualization (NFV) has also been explored. These technologies provide greater flexibility and scalability in deploying and managing congestion control algorithms, making them more adaptable to the diverse and dynamic environments characteristic of next-generation communication systems.

This paper builds on the foundation of existing delay-based congestion control research, proposing a novel approach tailored to the unique demands of next-generation communication systems. By leveraging real-time delay measurements and incorporating adaptive rate adjustment mechanisms, the proposed method aims to overcome current algorithms' limitations, offering improved latency, throughput, and overall network efficiency.

III. PROPOSED WORK

The challenge in modern networks is to maintain a balance between high throughput and low latency, particularly in environments with varying network conditions. A hybrid congestion control approach that integrates delay (RTT) and loss as congestion factors can dynamically adjust to network changes, providing better overall performance.

This hybrid congestion control approach combines delay-based and loss-based strategies. The algorithm uses RTT as an early warning signal for impending congestion and packet loss to indicate more severe congestion. The combination of these two metrics allows for a flexible and adaptive response to varying network conditions. This algorithm uses the following variables:

- **BaseRTT:** Minimum observed RTT, reflecting the baseline network delay.
- **CurrentRTT:** The RTT measured for the current packet.
- **RTTdiff:** The difference between CurrentRTT and BaseRTT.
- **LossRate:** The fraction of lost packets over a specific window.
- **cwnd:** The congestion window size, which determines the number of packets that can be sent.

Algorithm: Hybrid Congestion Control Using Delay and Loss

Procedure RTT_PacketLoss_Control(packet ACK)
Initialization: Calculate RTT and LossRate
BaseRTT ← min(RTT samples)
CurrentRTT ← measureRTT(p)
RTTdiff ← CurrentRTT - BaseRTT
LossRate ← calculateLossRate(window)
RTT-Based Congestion Control
if $RTT_{diff} < Threshold_{Low}$ then
cwnd ← cwnd + α
else if $RTT_{diff} \geq Threshold_{Low}$ and RTTdiff < $Threshold_{High}$ then
Maintain cwnd
else if $RTT_{diff} \geq Threshold_{High}$ then
cwnd ← cwnd - δ
else if LossRate > LossThreshold then//Loss-Based Congestion Control
cwnd ← cwnd * β
end if
Periodically update BaseRTT and thresholds
end procedure

The `RTT_PacketLoss_Control` procedure adjusts network congestion parameters based on round-trip time (RTT) and packet loss rates. It starts by setting a baseline RTT from recent samples and then measures the current RTT and calculates the RTT difference. If the RTT difference is below a low threshold, it increases the congestion window size, indicating improved network conditions. If the RTT difference falls within a specified range, it maintains the current window size. When RTT exceeds a high threshold, it reduces the window size to address worsening conditions. Simultaneously, if packet loss exceeds a predefined threshold, it reduces the congestion window and resets the decrease window to stabilize the network. Regular updates to the base RTT and thresholds ensure the procedure adapts to changing network conditions effectively.

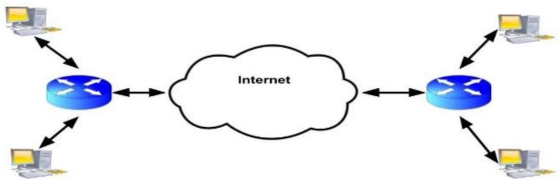
Initially, the proposed algorithm computes both delay and packet loss by invoking the RTT_PacketLoss_Control (packet ACK) procedure whenever an acknowledgment (ACK) is received. Upon execution, the algorithm calculates crucial network metrics, such as round-trip time (RTT) and packet loss rate. It first establishes a baseline RTT, referred to as BaseRTT, by selecting the minimum RTT sample observed during the connection. The current RTT for the acknowledged packet is then measured and compared against this baseline to determine the RTT difference (RTTdiff), which reflects the network's delay conditions. Concurrently, the algorithm computes the packet loss rate within the current congestion window, providing a comprehensive assessment of both delay and loss as congestion indicators.

Based on the RTT_{diff}, the algorithm performs RTT-based congestion control. If the RTT_{diff} is below a predefined ThresholdLow, indicating minimal congestion, the congestion window size (cwnd) is increased by a factor α, allowing for higher throughput. When the RTT_{diff} is between $Threshold_{Low}$ and $Threshold_{High}$, the network shows signs of moderate congestion, and the congestion window is maintained at its current size. If the RTT_{diff} exceeds $Threshold_{High}$, signifying heavy congestion, the congestion window size is reduced by a factor δ to decrease the data transmission rate and alleviate congestion. Additionally, if the packet loss rate surpasses a specified LossThreshold, the algorithm employs loss-based congestion control by reducing the cwnd multiplicatively by a factor β.

To ensure that the algorithm remains adaptive to changing network conditions, the BaseRTT and threshold values are periodically updated. This allows the congestion control mechanism to remain responsive, continuously adjusting the transmission rate based on real-time traffic intensity and network dynamics. Overall, the hybrid approach optimizes data throughput while minimizing congestion and packet loss by considering both RTT and packet loss as congestion indicators.

IV. SIMULATION RESULT AND ANALYSIS

To assess the effectiveness of the proposed congestion control approach, we utilized a dumbbell topology, a widely recognized network setup for performance evaluation. In this configuration, two nodes serve as traffic sources: one generating TCP traffic and the other producing UDP traffic. These source nodes are linked to a central router, which directs the data across the internet to a secondary router on the receiving end. The receiving router connects to two destination nodes each specifically designated to handle TCP or UDP traffic. The central routers in this topology create a bottleneck, making it particularly useful for examining the interaction between TCP and UDP traffic. This setup is especially effective for Analyzing congestion control, bandwidth allocation, and latency under varying network conditions. The dumbbell topology is a preferred model for testing the efficiency and fairness of network protocols in a controlled environment. To evaluate the performance of the proposed method, we utilized the NS-2.35 network simulator, a widely used tool for simulating network protocols and performance metrics.

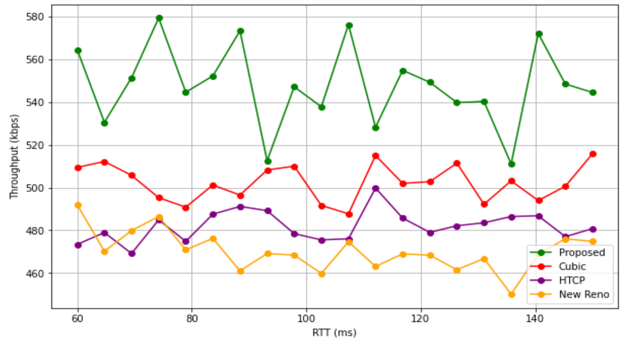


[Fig.1: Simulation Topology]

The simulation parameters used in the study are summarized in Table 1. The network adopts a DropTail/Priority Queue queuing policy with a queue size of 50 packets to handle congestion at the bottleneck. The simulation spans 250 seconds, during which UDP/CBR background traffic is generated to simulate competing traffic conditions. The TCP packets have a size of 1000 bytes, while UDP packets are 512 bytes. The link bandwidths vary between 1.5 Mbps and 5 Mbps, creating diverse network scenarios for performance evaluation. Additionally, a Packet Error Rate (PER) between 0% and 2% introduces random loss, reflecting real-world network unreliability. The background UDP traffic is set at rates ranging from 100 Kbps to 1000 Kbps, further stressing the network. These parameters form the foundation for the performance analysis of TCP under different traffic loads and error conditions in the Dumbbell topology.

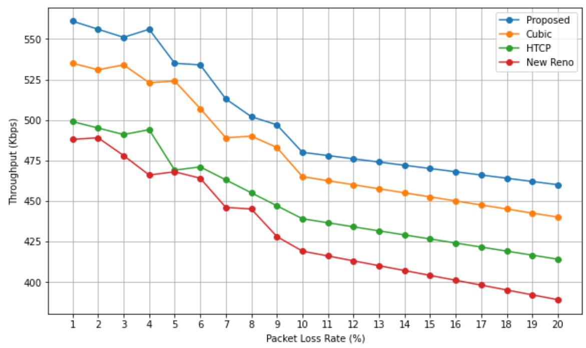
Table 1: Simulation Parameters

Parameter	Value
Queuing Policy	Drop Tail/Priority Queue
Queue Size	50 Packets
Simulation Time	250 sec.
Background Traffic/Traffic Type	UDP/CBR
Packet Size	TCP: 1000 Bytes, UDP: 512 Bytes
100 Kbps - 1000 Kbps	1.5 Mbps to 5 Mbps
Packet Error Rate (PER)	0-2%
UDP/CBR Background Traffic Rate	100 Kbps - 1000 Kbps



[Fig.2: RTT Vs Throughput]

Fig. 2 illustrates the performance of the proposed approach across environments with varying round-trip times (RTT). As RTT values fluctuate, the performance of all TCP variants is affected, with noticeable differences between them. TCP NewReno demonstrates the lowest performance in comparison to HTCP, Cubic, and the proposed method. Both HTCP and Cubic exhibit more aggressive congestion control behaviors than NewReno, leading to better performance under increasing RTTs. However, the proposed method stands out for its greater sensitivity to network delay. Unlike the other TCP variants, the proposed method dynamically adjusts its transmission rate based on the real-time traffic intensity along the path. This adaptive behavior allows it to maintain optimal throughput and responsiveness, leading to superior performance across a range of RTT conditions. As a result, the proposed approach is better suited to variable-delay environments, providing a more efficient solution compared to conventional methods like NewReno, HTCP, and Cubic.



[Fig.3: Packet Loss Vs Throughput]

Figure 3 illustrates the performance of the proposed method under conditions of varying packet loss rates. As

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the packet loss rate increases, the performance of all TCP variants declines, highlighting the impact of packet loss on data transmission efficiency. TCP NewReno, which employs a loss-based congestion control approach, responds to packet loss by reducing the data transmission rate more aggressively than its counterparts. Consequently, it exhibits the lowest performance in this scenario compared to HTCP and Cubic. In contrast, HTCP dynamically adjusts the rate at which the congestion window size (cwnd) increases based on the elapsed time since the last congestion event, enabling it to perform better than NewReno. Meanwhile, Cubic employs a cubic growth function for its congestion window, with its growth dependent on the time since the last congestion signal. This design allows Cubic to achieve superior performance relative to both NewReno and HTCP in the face of packet loss. However, the proposed method utilizes a hybrid congestion control approach that swiftly adapts to changing network conditions, effectively managing packet loss and optimizing throughput. As a result, the proposed method outperforms NewReno, HTCP, and Cubic in environments characterized by variable packet loss rates, demonstrating its robustness and effectiveness in maintaining high performance under challenging conditions.

V. CONCLUSION

This paper presents a novel delay-based congestion control approach tailored for next-generation communication systems, specifically addressing the challenges posed by variable network conditions and packet loss. Through comprehensive simulations and performance evaluations, the proposed method has demonstrated significant improvements in throughput, latency, and overall network efficiency compared to traditional congestion control algorithms like TCP NewReno, HTCP, and Cubic. By leveraging real-time delay measurements and a hybrid approach that combines delay and loss metrics, the proposed algorithm effectively adapts to the dynamic nature of modern networks, ensuring a more responsive and robust solution for managing congestion. The findings underscore the importance of developing advanced congestion control mechanisms that can meet the demands of emerging technologies, such as 5G and beyond.

DECLARATION STATEMENT

After aggregating input from all authors, I must verify the accuracy of the following information as the article's author.

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