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Context-adaptive and QoS-guaranteed flow scheduling optimization in multipath multimedia transmission over MPTCP

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Abstract
This paper presents a context-adaptive QoS optimization solution designed for Multipath-TCP (MPTCP) control layer, in multimedia transmission environment. To improve the QoS of multimedia transmission, we jointly consider path scheduler of MPTCP to reduce congestion based on context metrics and a global QoS guarantee approach adjusting tunable context metrics under the analysis of the relationship between QoS metrics and contexts. Then we simulate the shared bottleneck experiment, compare the performance of the algorithm we proposed with existing algorithms, and verify the good performance of our algorithm in improving the throughput while ensuring the flow fairness. Our approach is different from traditional congestion control algorithm based on packet loss, and it tells whether the packet loss is due to congestion or other reasons, adjusting the congestion window size more adaptively.

Keywords QoS · MPTCP · Multimedia transmission

1 Introduction

With the rapid development of multimedia technology and network communication technology, numerous applications in various domains such as entertainment, education, business, medical service and advertising applied multimedia communication technologies to achieve better user experience. According to the Cisco Visual Networking Index (VNI), global consumer Internet video traffic is 28,768 petabytes (PB) per month in 2015, and grow to 109,907 PB per month in 2020 [1]. Internet video streaming and downloads are about to occupy a larger share of bandwidth, which will increase to more than 80 percent of all users' Internet traffic [2]. Despite the increasing popularity of research of multimedia transmission, a related worldwide study shows that stalls and buffers still trouble the service providers and users [3]. Additionally, video Peak Signal-to-Noise Ratio (PSNR), startup delay and stability are also impairments to user experience. The degradation in the contracted QoS is often unavoidable, thus there is a need to provide real-time QoS monitoring that not only is capable of monitoring the QoS support in the network but that can also take actions in real-time manner to sustain an acceptable multimedia presentation quality when the QoS level degrades.

Currently, transport layer protocol used in transmission of multimedia traffic in Internet is TCP. As more and more endpoints in the Internet have multiple network adapters, traditional TCP cannot take full advantage of these resources. Multipath-TCP (MPTCP) [4] is considered to be an effective transport layer technology which meets specific quality of service (QoS) requirement of the real-time service and realize optimal allocation of network resources simultaneously. It is proposed by Internet Engineering Task Force (IETF) and designed to achieve the goal of better performance and robustness over today's network. The idea of applying MPTCP to multimedia transmission is attractive as it can increase the capacity of network and realize seamless switch in general cases. However, due to out-of-order packet delivery, conventional MPTCP methods still suffer a bottleneck in the data-sorting and the degradation of total throughput when round-trip time of paths in the heterogeneous network are considerably different. As argued in [5], the authors analyze the impact of MPTCP on DASH video streaming and draw the conclusion that even though MPTCP is beneficial when the bandwidth is constant, it is sensitive to bandwidth oscillation. In the heterogeneous multimedia

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transmission environment over MPTCP, networking characteristics are divergent between paths, which definitely bring out undesirable degradation of QoS in terms of the times and duration of stall, long startup delay, unsteadiness and other bad experiences. But there are some challenging issues for the QoS guarantee in multimedia transmission, especially in multimedia streaming which has real-time demand. As the analysis in [6, 7], the tunable context variables are limited. Furthermore, due to the casual and time-varying relationships between QoS metrics and context, it is difficult to observe the motive of the service exception (Fig. 1).

In conclusion, we have following contributions:

- We propose a context-adaptive QoS optimization solution framework based on details of MPTCP and features of multimedia data. MPTCP packets will not be blocked by middle-boxes due to this layer, which makes it perform more effective. To achieve a high-quality multimedia transmission service, QoS can be guaranteed by a pre-dominant congestion control scheme.
- Based on the proposed framework, we present a QoS-oriented MPTCP path scheduler to distribute data efficiently. We jointly consider path scheduler of MPTCP used to reduce congestion based on context metrics and a global QoS guarantee approach adjusting tunable context metrics under the analysis of relationships between QoS metrics and contexts.
- We propose an approach to find optimal context adaption strategy by analyzing and exploiting the relationship between expected QoS metrics and homologous contexts.

The remainder of the paper is organized as follows: we discuss some related works in Sect. 2. In Sect. 3, we introduce MPTCP architecture in brief. Then the QoS optimized solution is presented in Sect. 4 and validation and evaluation in Sect. 5. In the end, we conclude the paper in Sect. 6.

2 Related work

2.1 MPTCP path scheduler

Because there are additional path options, a thoughtful path schedule schemes is responsible for the data distribution and achievement of greater total throughput while an unwise one might introduce congestion of some paths or higher end-to-end delay according to the data carried by the path with larger loss rate. Conventional MPTCP chooses the path with lowest smoothed round trip time (sRTT) to assign current data packet, which does not take into account path heterogeneity. [8] proposed the ATLB algorithm combining a variety of the characteristics to score every available path and selecting the path with highest score. Path congestion is taken into account in [8]. M. Becke proposed an approach of confluent sequence numbering [9]. In [10], authors proposed a new scheduling strategy which reduces the jitter by sending out-of-order packets in different sub-streams based on present queuing time and sRTT in different sub-paths so that data packets can arrive at the receiver in order with decreased delay.

2.2 QoS guarantee

In the aspect of QoS guarantee, especially in multimedia streaming with low loss-tolerance and jitter-tolerance, traditional approaches are based on rules which demand definite borders of QoS parameters. To overcome inflexibility and ineffectiveness of the obsolescent methods, some adaptive QoS guarantee solutions have been proposed. [11] provided an energy-efficient QoS-aware routing algorithm for Wireless Multimedia Sensor Networks (WMSN) considering limited resources of sensors and energy consumption simultaneously. The authors of [12] proposed another energy-aware packet transport mechanism to realize QoS. Beacon messages which carry the position information are utilized to implement the method. Either multi-path or time-constrained routing is considered in Reliable, Real-time Routing protocol (3R) [13]. Suleiman Y. Yerima and Khalid Al-Begain proposed a dynamic buffer management scheme to guarantee the end-to-end QoS of the real-time streaming flow, whilst protecting the non real-time flow from starvation [14].

3 MPTCP architecture

MPTCP is an extension of TCP which allows concurrent usage of multi-interface in data transmission. It is TCP-friendly, namely a MPTCP-enabled endpoint, that can communicate with an endpoint without MPTCP supported by TCP with no modification in application layer. MPTCP architecture is depicted in Fig. 2. To achieve multipath transmission, it divides transport layer into several sub-layers. MPTCP control level, which is a semantic layer instantiation orienting to application layer, is responsible for MPTCP connection control. Besides, network-oriented sub-flow level aims at ensuring protocol's compatibility with Internet. Fortunately, MPTCP packets will not be blocked by middle-boxes due to this layer, which makes it perform more effective than SCTP.

To achieve a high-quality multimedia transmission service, QoS can be guaranteed by a predominant congestion control scheme. MPTCP congestion control scheme should satisfy three goals designed in IEEE RFC 6356[21]:

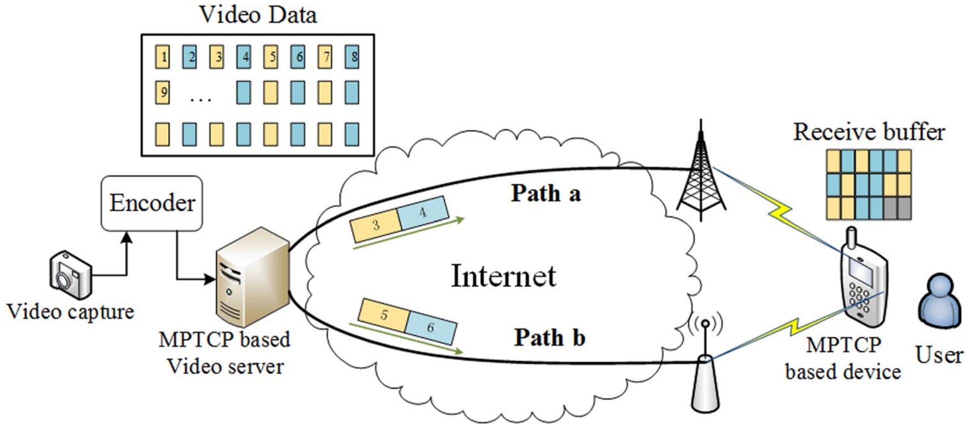

Fig. 1 Multipath Multimedia Transmission over MPTCP

a) MPTCP's total throughput should be at least as high as a single path flow; b) in shared-bottleneck links, MPTCP should not occupy more capacity than if it was a single flow; c) subject to satisfying the former two conditions, MPTCP has to move the traffic of congested paths to uncongested paths. Our algorithm is designed for MPTCP Control layer, which is handling the problem that which sub-flow will be chosen to carry out the current traffic.

Individual subflows in MPTCP use standard TCP signaling. However, MPTCP incorporates a Data Sequence Signal (DSS) into the connection to provide an overview of the data flow across all TCP subflows within the session. The sender's sequence numbers include both the overall data sequence number and the subflow sequence number, which is used to map the data segment to a specific subflow. The DSS Data ACK sequence number represents the cumulative acknowledgment of the highest in-order data received by the receiver. MPTCP does not utilize Selective Acknowledgment (SACK), as this is delegated to the individual subflows. If data loss occurs and blocks a particular subflow, the sender can retransmit the data through additional subflows to prevent congestion. Each subflow operates using a conventional TCP sequencing algorithm, so an unreliable connection may cause a subflow to become inactive. In such cases, MPTCP can employ alternative subflows to retransmit the data, and if the issue persists, it can reset the affected subflow using a TCP Reset within that subflow's context.

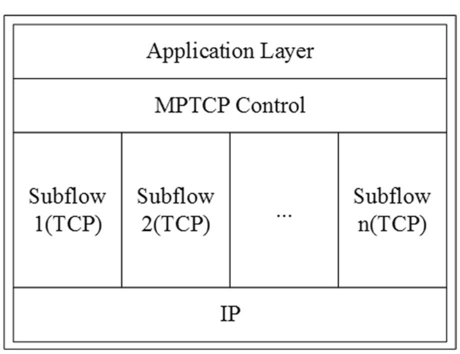

Fig. 2 MPTCP Architecture

4 Proposed QoS guarantee method

We propose to jointly consider path scheduler of MPTCP used to reduce congestion based on context metrics and a global QoS guarantee approach adjusting tunable context metrics under the analysis of relationships between QoS metrics and contexts.

4.1 QoS-oriented MPTCP path scheduler

Throughout this section, we consider a scenario in which two paths between a source endpoint and a destination endpoint with different characteristics are available.

4.1.1 Problem analysis

Our algorithm is aiming to improve QoS in terms of total throughput when a multimedia transmission happens. Out-of-order arrival is one of factors decreasing total throughput. Out-of-order packages cost large buffer resource and occupy part of CPU processing time for rearrangement. Traditional TCP discards these packages to save the buffer and CPU time, which is not useful in MPTCP since ACK messages have been sent to source on the sub-flow yet when packages arrive correctly. It is also unwise to delay the ACK messages so that timeout events will be activated and size of send window will be decreased in the source. The reason of out-of-order arrival is heterogeneity and randomness of paths primarily presented as delay and congestion. Delay metrics include RTT, bandwidth, loss rate, the size of send window and receive window, and the size of congestion window can measure degree of congestion. A package, transported in the fastest path without causing loss and congestion, will be delivered to destination earlier.

4.1.2 Algorithm description

We propose to jointly consider sRTT (smooth round-trip time) and congestion situation. According to [15, 16], we keep a two-level structure of send queue to implement our algorithm so that packets can be queued in a cautiously allocated order. The structure is shown in Fig. 3. Furthermore, we refer to the configuration of single sub-flow's send buffer as [15, 17, 22, 23] proposed, which can make more available scheduling via several subflow send queues avoiding single queue failure.

For sub-path i, a score S_i will be assigned as Formulation 1 to measure performance of every path as

$$S_i(t) = \frac{num\_send\_imi_i(t)/cwnd_i(t)}{delay_i(t)} \tag{1}$$

where $delay_i$ denotes estimated the expectation ofcurrent transmission's time interval from scheduled to arrival or loss, meanwhile $num\_send\_imi_i(t)$ denotes the number of packets what will be sent immediately on path i. Larger $num\_send\_imi_i(t)$ means the less congested path. Let $queuing_i$ be the number of scheduled packets without sending and $ppgn\_delay_i$ be the propagation delay on path i, we define $delay_i$ as

$$\begin{aligned} delay_i(t) \approx \lfloor \frac{queuing_i(t)}{cwnd_i(t)} \rfloor &\times [lr_i(t) \times t_{timeout} + (1 - lr_i(t)) \\ &\times sRTT_i(t)] + [lr_i(t) \times t_{timeout} + (1 - lr_i(t)) \\ &\times ppgn\_delay_i(t)] \end{aligned} \tag{2}$$

where $queuing_i$ indicates the number of packets waiting in the send buffer like the cube with diagonal in Fig. 4, and $\lfloor \frac{queuing_i(t)}{cwnd_i(t)} \rfloor$ indicates how much end-to-end delay should be waited when current packet is ready to be transmitted. The first term indicates waiting time and the second term represents delivery time considering loss rate of path i. We assume $ppgn\_delay_i(t) = sRTT_i(t)/2$ to simplify the problem.

While estimating lr_i, we keep a dynamic observation duration $d_i(t)$. As we know, a loss will lead to jitter. Assuming the observed round trip time values on path i are $\{rtt_1, rtt_2, \dots, rtt_m\}$, and jitter $J_i(t)$ can be calculated as

$$J_i(t) = \frac{(rtt_{curr} - rtt_{avg})}{(rtt_{max} - rtt_{min})} \tag{3}$$

where $rtt_{avg} = \frac{1}{m}\sum_{i=1}^{m} rtt_i$. $d_i(t)$ will be updated as

$$d_i(t) = d_i(t-1) + d_i(t-1) \times J_i(t) \tag{4}$$

So, if $J_i(t) > 0$ or $J_i(t)$ is large enough namely the network congestion will become heavier and this will increase the possibility of packet loss. According to the update formula (4), the observation duration will be extended and vice versa. We can introduce jitter factors to dynamically adjust the observation duration of packet loss, and we can amend the estimated duration of packet loss based on current network congestion. $loss\_rate_i$ can be computed as the ratio of number of loss packets to number of total packets during the observation.

After dynamic evaluation, we choose the path with the largest score as carrier and assign the path with a data packet of the length that equals the congestion window. According to $queuing_i(t) \times delay_i(t-1)$, we can see that the delay of current packet determines whether the current packet can arrive at the receiver on time and influences the delay of the upcoming packet.

As Fig. 5 shows, if sub-path terminates connection or the path is invalid for some reason during the data transmission of MPTCP, re-scheduling strategy will be applied. Assume the sub-flow i now can't be used, re-scheduling will put the data packet in the private buffering queue of sub-flow i into the shared queue in sequence and keep the sequence of data

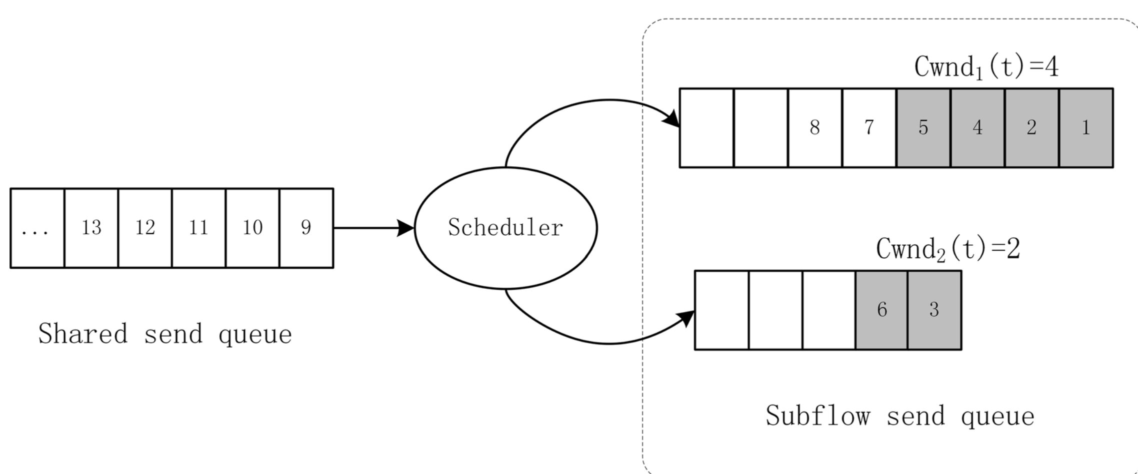

Fig. 3 Send Queue Configuration

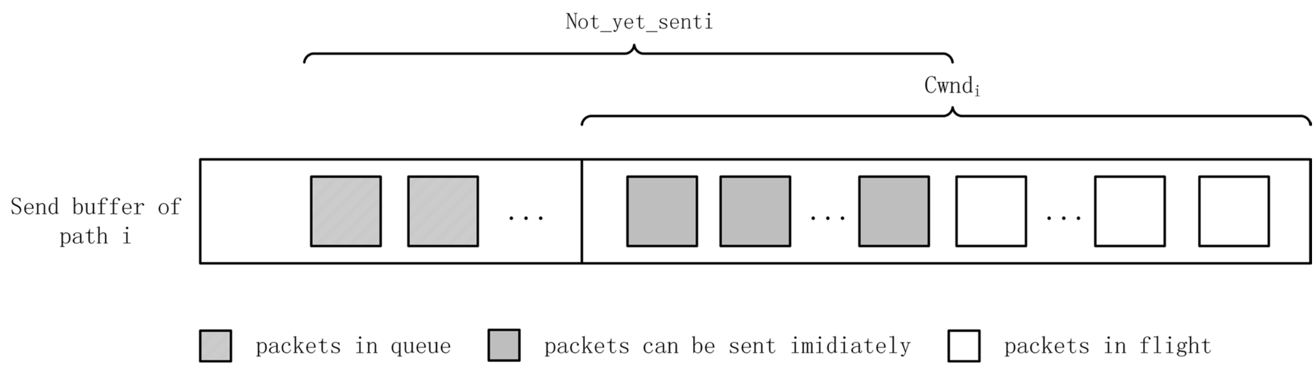

Fig. 4 Single Sub-flow's Send Buffer Configuration

packet monotonic from queue head to queue tail, awaiting to be scheduled to other sub-flows.

4.2 Global QoS guarantee

4.2.1 Problem analysis

QoS metric varies with the corresponding context variables[18]. For example, loss rate, a QoS metric, which may bring mosaics and stalls in a real-time streaming system with a relative large value, is commonly caused by a context–network congestion [19, 20, 24]. A critical issue in this scenario is how to make QoS guarantee as automatic as possible. To this aim, a context acquisition system can be used in numerous stream management with regards to the QoS, to adaption to the available bandwidth, and to the capacity of the involved devices. Consequently, an effective scheme of QoS guarantees what tunes some context to appropriate values, is absolutely necessary. In the multimedia transmission frame over MPTCP we propose, factors signal strength, available bandwidth, current throughput, congestion situation, data rate, etc. can be considered as context which has potential impact on QoS of the transmission. Some of them are adjustable and some not. We should dig out and quantize the relationships among contexts and relationship between contexts and QoS metrics, so that the optimal context values can be computed by given QoS metrics. Procedures of our method is shown in Fig. 6.

4.2.2 Context discretization

As we know, contexts are gathered from different sources and applications. Therefore, they may be continuous or discrete. Our previous work in [6] illustrated that the result of causal relationships using raw continuous context variables learned by Bayes Network (BN) are redundant or incorrect. Hence, a discretization process is imperative to amend the learning procedure.

Fuzzy set is proved to be a generally prevalent method of discretization. What allows gradual evaluation of the

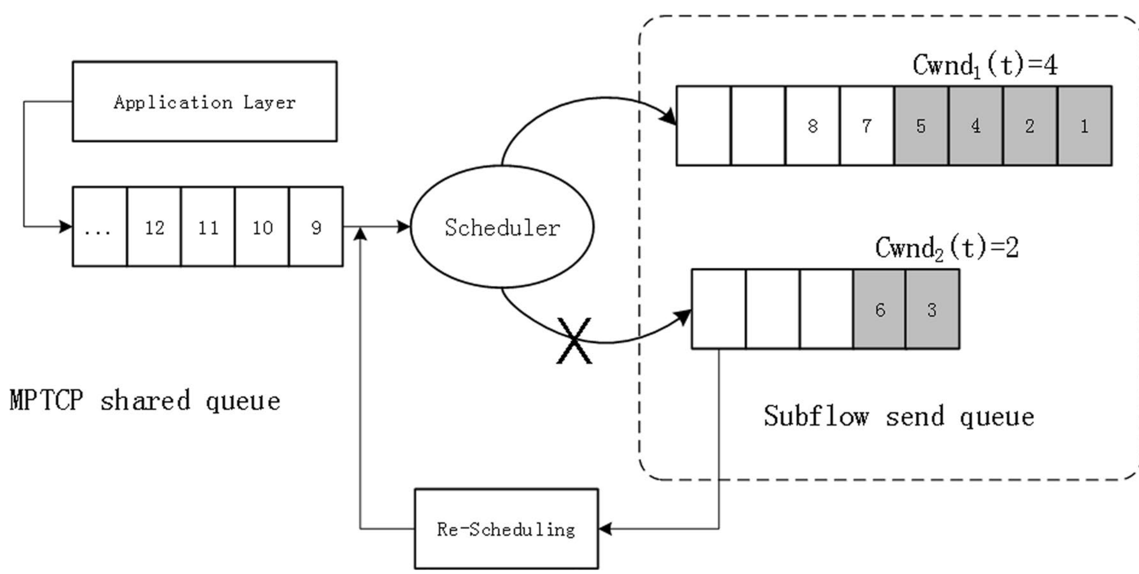

Fig. 5 Re-scheduling Strategy

membership of elements in a set is a membership function $f_A(\chi)$ valued in the real unit interval [0, 1] in fuzzy set A. However, this function is too subjective to define and the homologous discretization results may be distinct from different designs. So we quote the method of our previous work in, a novel self-adaptive approach to discretize continuous context.

Algorithm 1 Context Discretization

Input: Continuous sample set Ω of a context/QoS metric
Output: Discrete value set Ω_D of Ω
Initialize sets $\Omega_D \leftarrow \Omega, V \leftarrow \varnothing, F \leftarrow \varnothing$
$(V, F) \leftarrow$ MEMFUNC_ACQ(Ω)
for all sample s in Ω **do**
    $dv_s \leftarrow \max\{f_v(s) | f \in F, v \in V\}$
    Ω.add(dv_s)
end for
return Ω_D

function MEMFUNC_ACQ(Ω)
    $PD \leftarrow K - DENSITY(\Omega)$
    $\{GAUSSIAN\_FUNC, RMSE\} \leftarrow GAUSSIAN\_FITTING(PD)$
    $G \leftarrow GAUSSIAN\_FUNC_{argmin}\{RMSE_i | RMSE_i \in RMSE\}$
    $a_{max} \leftarrow \max\{a_i | a_i$ is the coefficient of g for g in $G\}$
    $j \leftarrow 0$
    for all g in G **do**
        $j \leftarrow j + 1$
        $F[j] \leftarrow g$
        $V[j] \leftarrow cng$
    end for
    for all f in F **do**
        $f \leftarrow \frac{f}{a_{max}}$
    end for
return $\{V, F\}$
end function

According to the previous work, a few contexts and QoS metrics obey approximate Gaussian distribution and most of them obey a summation of Gaussian distributions. In our research, suppose continuous context C has n values after discretization. Thus the probability density $f(\chi)$ of an approximate Gaussian distribution, which has n terms and each of them represent a unique characteristic, can be defined as:

$$f(\chi) = \sum_{i=1}^{n} g_i(\chi) \tag{5}$$

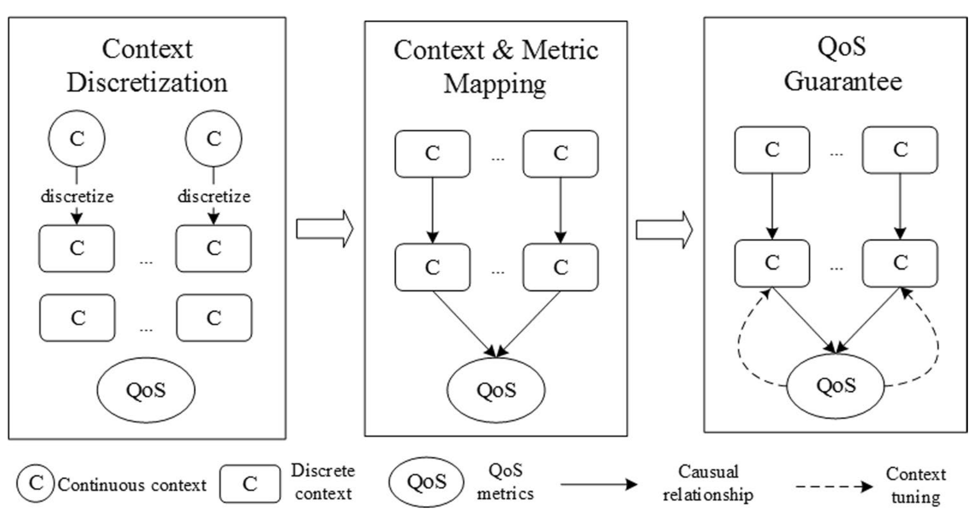

Fig. 6 Three Procedures of the Proposed Approach

where n equals to the number of available samples; $g_i(\chi)$ is the membership function mapping to a discretized value DC_i, which is defined as:

$$g_i(\chi) = k_i \times exp\left(- \left(\frac{\chi - \mu}{\sigma_i}\right)^2\right) \tag{6}$$

where k_i is the coefficient and is normalized by dividing maximum of set $\{k_i|i \in [1,n]\}$; μ_i and σ_i are the expectation and standard deviation of the ith sample. We choose root-mean-square-error (RMSE) as lost function in Gaussian fitting. The details of context discretization algorithm is discussed in Algorithm 1, where procedure $Memfunc\_acq(\Omega)$ is used to obtain membership functions and discretization results of context sample set Ω.

4.2.3 Context of multipath and QoS metric mapping

Mapping Context of multipath to QoS metric achieves to dig out the causal relationship between them in a real-time manner. Our previous research in [6] has proved that Bayesian network is practical in learning and modeling these causal relationship. In this paper, we extend the Bayesian network built on single-path transmission to multipath transmission.

Formally, Bayesian network is a directed acyclic graph (DAG) model based on probability reasoning, whose nodes represent a set of random variables and arcs denote conditional dependencies of nodes. If node A is the parent of node B, the variable B indicates must be the result of the one A represents. Assuming that there are two paths in our MPTCP multimedia transmission system, we build the BN's structure for the context and QoS metrics. Firstly, we divide context metrics into two groups: primary-group and runtime-group. Primary group include available bandwidth of each path and video streaming rate of sender end, which is global and will be settled after application starts up. They should be the parent of all the rest nodes as the original reasons. Average end-to-end delay and loss rate of every path form the runtime context group, because they are sequentially configured or loaded and are tightly coupled local contexts. Furthermore, average video PSNR and playback buffer size are elected as QoS metrics which is the leaf node of the DAG. Figure 7 shows the network structure of our model and algorithm describe the mapping process step by step.

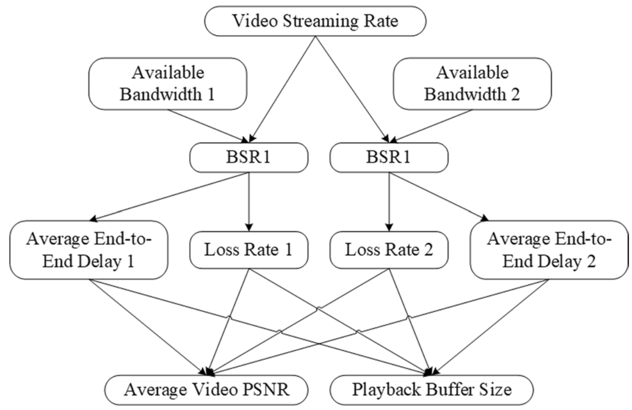

Fig. 7 Bayesian Network Structure for the Context and QoS Metric Mapping

5 Experimental evaluations

5.1 Experiment settings

To verify the effectiveness of the optimization algorithm, we simulate the shared bottleneck experiment as shown in Fig. 8. With the experimental settings in [9], Node A establishes a single-path TCP, node B establishes a MPTCP connection including two sub-flows, and the three paths share the same bottleneck path. The network of the bottleneck path is 10.1.1.0 and the IP address of each network port is shown in Fig. 8.

5.2 Network parameters

In Fig. 8, three paths share the same bottleneck path and the bandwidth of the bottleneck path is set to 10 Mbps, delay is 10 milliseconds, packet loss rate is 0.1%. When the three paths transmit at the same time, we need to guarantee the fairness of congestion control algorithms by preventing them from invading other paths.

5.3 Evaluation index of performance

To evaluate the MP-CDG congestion control algorithm, we mainly consider fairness and congestion equilibrium, and this is the explanation of the two indexes.

5.3.1 Fairness

Fairness includes network fairness and bottleneck fairness. Network fairness means that the MPTCP connection and the single-path TCP connection in the network should get the same throughput in the network. Network fairness can ensure the use of multi-path, and can also ensure that the use of multi-path transmission does not harm the single-path TCP flow. It can be formulated as:

$$\sum_{r\in R} \frac{\hat{w}_r}{rtt_r} = \max_{r\in R} \frac{\hat{w}_r^{TCP}}{rtt_r} \tag{7}$$

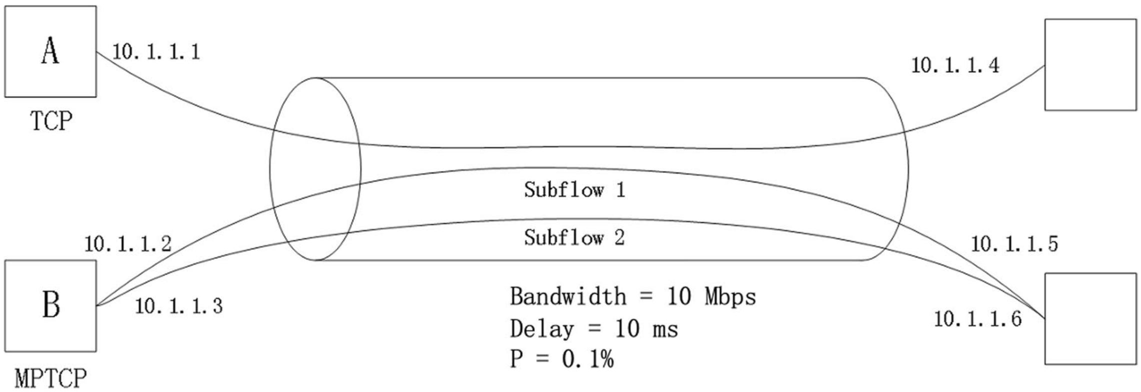

Fig. 8 MP-CDG Algorithm Simulation Network

where R denotes the sub-flow set, $\hat{w}_r$ represents the steady-state value of the sub-flow r's congestion window in the congestion avoidance phase, $\hat{w}_r^{TCP}$ represents the steady-state value of the single-path TCP flow r's congestion window in the congestion avoidance phase, and rtt_r is the round-trip time of the path r.

The significance of bottleneck fairness in the MPTCP scenario is that the throughput of each sub-flow is reduced to T/n, where T is the throughput that a regular TCP flow obtains on this bottleneck link, and n refers to the total number of sub-flows.

5.3.2 Congestion equilibrium

The purpose of congestion equilibrium is to obtain a resource pool. The verification of the resource pool is mainly measured in two aspects. First, whether the algorithm can effectively average the throughput of each connection; second, whether the algorithm can relieve connection congestion.

5.4 Fairness evaluation of different algorithms

In the NS-3 simulation platform, the Multipath Congestion Detection and Guided algorithm (MP-CDG) is implemented, along with the Link Improvement-based adaptation (LIA) algorithm and the Uncoupled algorithm. These algorithms are used to optimize network performance. The simulation scenario depicted in Fig. 8 is utilized to test the functionality and effectiveness of these algorithms in the NS-3 environment.

Table 1 shows the data comparison of the average throughput of Node A and B. When using LIA algorithm, the average throughput of Node A is 4.4 Mbps, and the average throughput of Node B is 4.5 Mbps. The simulation results with LIA algorithm can be considered as the baseline for the comparison with other algorithms. When using the MP-CDG congestion control algorithm, the average throughput of node A, the single-path TCP, is 4.4 Mbps, and the average throughput of node B, MPTCP connection, is 4.8 Mbps. Additionally, when using the Uncoupled algorithm, the average throughput of Node A and B is 3.1 Mbps and 6.6 Mbps, respectively. As shown in Fig. 9, the simulation results also show the performance comparison intuitively. Therefore, MP-CDG algorithm can improve the average throughput of Node B and ensure the fairness that the average throughput of node A is not affected. Although the average throughput of node Bcan be improved in Uncoupled algorithm, the average throughout of node A reduced, especially compared with the result in the baseline.

Table 1 Average Throughput Comparison

Algorithm	MP-CDG	LIA	Uncoupled
Node A	4.4 Mbps	4.4 Mbps	3.1 Mbps
Node B	4.8 Mbps	4.5 Mbps	6.6 Mbps

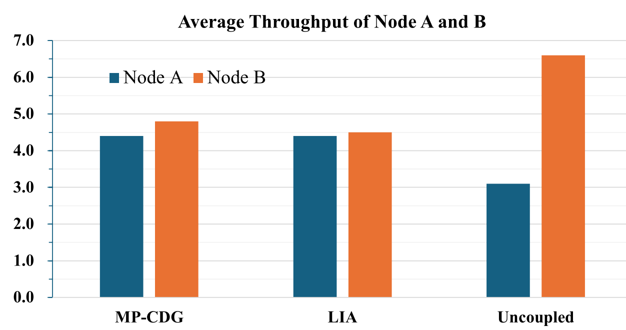

Fig. 9 Throughput Comparison of Different Nodes with Different Congestion Control Algorithms

5.5 Congestion evaluation of different algorithms

As shown in table 2, the evaluation results reveal significant variations in the performance of different congestion control algorithms. Through detailed analysis of these results, MP-CDG algorithm gains the insights into the trade-offs inherent in its suitability for specific network environments. Figure 10 shows the throughput changes of different congestion control algorithms. The performance of MP-CDG algorithm can achieve the throughput improvement compared with LIA algorithm, and can ensure the fairness compared with Uncopuled algorithm.

6 Conclusion and future work

Multi-path parallel transmission has the advantages of high speed, high throughput, and high bandwidth utilization. Currently, there are multi-path parallel transmission related technologies in the application layer, transport layer, and network layer. In this paper, the multi-path parallel transmission technology is researched based on MPTCP protocol. The MPTCP protocol is analyzed from the point of view of congestion control algorithm and data scheduling algorithm, and the improvement suggestions are put forward. The context-adaptive QoS optimization algorithm is proposed.

The context-adaptive congestion control algorithm is different from the traditional packet loss-based congestion control algorithm, and it is based on delay. It will decide whether the packet lost is caused by network congestion. Only when the packet loss occurs and the sending sub-path load is equal to or equal to the network capacity, is packet loss considered to be a sign of network congestion, and then the sub-flow congestion window will be attenuated multiplicatively. The algorithm also establishes an RTT back-off mechanism for each sub-stream. When the RTT continues to rise but packet loss does not occur, the transmission rate is gradually reduced with a certain back-off probability. Through simulation experiments, the algorithm is compared with LIA and Uncoupled congestion control algorithms, and tested under the bottleneck link scenario. It is found that the algorithm can effectively maintain the fairness when sharing the bottleneck with the TCP, and can obtain higher throughput than LIA algorithm, achieving load balance. Although the Uncoupled algorithm achieves the highest throughput, it exhibits aggression against TCP connections and does not meet the design goal of fairness.

The improvement of the MPTCP protocol in this paper is based on the three goals of the MPTCP design and proposes the congestion control algorithm and data scheduling algorithm. Although the algorithm proposed in this paper reaches a certain degree of effect, due to the complexity of the real network, other parts of the MPTCP mechanism are not optimal and must be further studied and improved.

Table 2 Average Throughput Comparison (bits)

Time (ms)	10	20	30	40	50	60	70	80
MP-CDG	1002	1502	952	1286	1476	998	1022	1833
LIA	205	1087	208	186	1099	308	104	1305
Uncouple	1901	2387	2206	2001	2288	2198	2407	3902

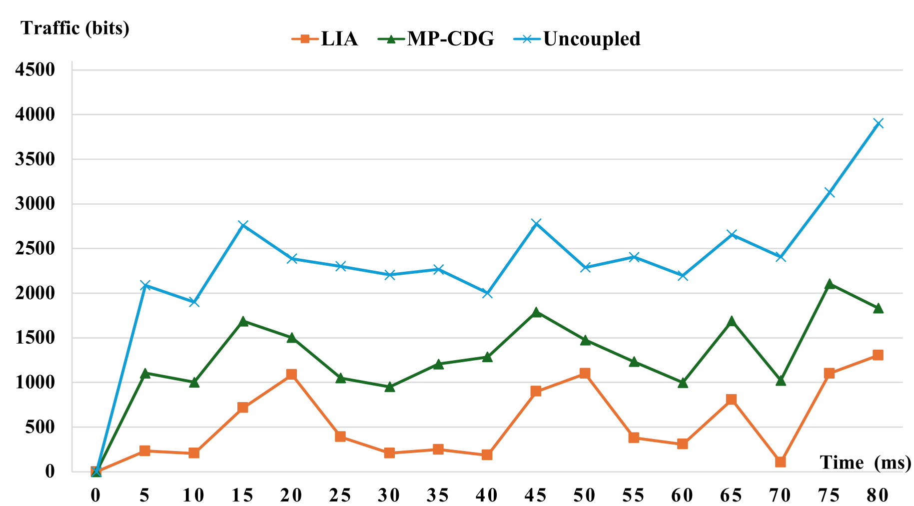

Fig. 10 Throughput Changes of Different Congestion Control Algorithms

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