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Receiver-Assisted Partial-Reliable Multimedia Multipathing Over Multi-Homed Wireless Networks

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ABSTRACT With the advancement of multimedia technologies and driven by the ever-increasing user interests in the variety of multimedia applications, the content-rich multimedia streaming services tend to be the most attractive service in future Internet. The multipath TCP (MPTCP), which uses multiple paths for parallel transmission and bandwidth aggregation, is considered to be the most potential transfer mechanism to satisfy the specific requirements of multimedia transmission in a multi-homed wireless network environment. However, the fully-reliable transmission nature of MPTCP can cause an unnecessary retransmission of expired multimedia data, block message from handing over to upper layer, and thus degrade the user-perceived multimedia service quality. In this paper, we propose a receiver-centric partial-reliable multipath transport solution (referred to as *recPR-MPTCP*) to support real-time Internet multimedia applications. In the *recPR-MPTCP*, a partial reliability extension is presented which runs at receiver side in order to offer a partial reliability multipathing service for multimedia applications. In addition, a new one-way delay-based bandwidth aggregation method is included into the *recPR-MPTCP* in order to reduce the receive buffer blocking problem in MPTCP and increase multimedia services performance. Simulation results show that *recPR-MPTCP* outperforms the current MPTCP solutions in terms of throughput performance and user-perceived quality of service.

INDEX TERMS Multi-homed wireless networks, multimedia communication, multipath TCP, partial reliable, bandwidth aggregation.

I. INTRODUCTION

With the widespread availability of wireless networking technologies and driven by the ever-increasing user interests in the variety of multimedia applications, the content-rich multimedia traffic tend to account for an overwhelming majority of the world's internet traffic [1], [2]. According to a recent Cisco Visual Networking Index Complete Forecast [3], there will be more than 2 billion Internet multimedia/video users, and a massive 85% of the global Internet traffic will be multimedia/video by 2022. We have reasons to believe that, the high-quality interactive multimedia applications (e.g., mobile audio/video conference, massively multiplayer

online games, and so on) will become the most attractive service in future Internet [4], [5]. In the meantime, promoted by the advances in various wireless broadband access technologies, modern mobile devices are increasingly being attached more than one wireless network interfaces [6]. These devices can make use of multiple network links to satisfy the specific bandwidth requirements of multimedia transmission in a multi-homed wireless network environment, supported by Multipath TCP (MPTCP) technology [7].

The MPTCP is an extended version of TCP that supports the simultaneous use of several available interfaces for parallel transmission over multiple access links and bandwidth aggregation. Fig. 1 presents a typical MPTCP usage scenario in which two endpoints (a mobile device and a multimedia server) are connected with each other via two paths

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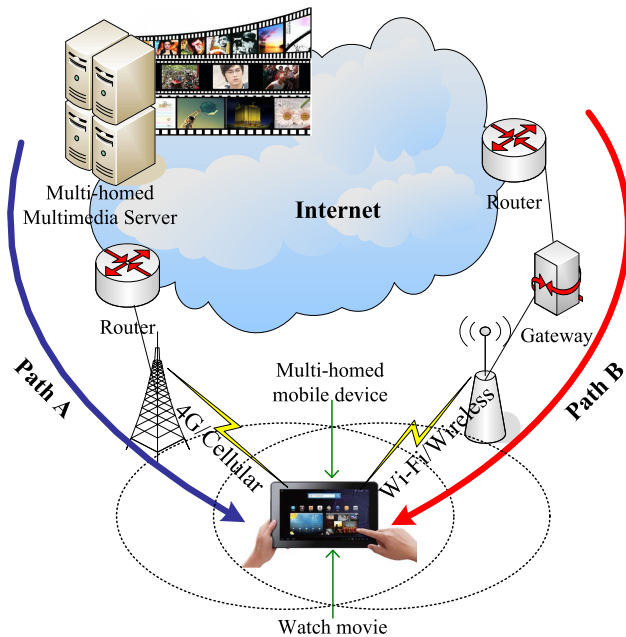


FIGURE 1. A typical MPTCP usage for wireless multimedia communication.

(A and B) simultaneously. The mobile device can aggregate the bandwidth of both path A (Wi-Fi connection) and path B (cellular connection) to download media content from the server. Such MPTCP-enabled multipath communication feature has a potential for providing the communication system with many and attractive benefits including bandwidth aggregation, goodput improvement, robustness enhancement. Moreover, as an emerging transport layer protocol, MPTCP inherits from TCP's socket APIs that are used by the most current network applications, which means MPTCP is able to be backward compatible with today's network infrastructure and does not require any changing/adding operation to the applications during the connection and communication process [8]. For these reasons, MPTCP is now considered one of the most promising transmission technologies which will be widely used in the future Internet environment [9], [10].

Although MPTCP towards transport-layer multipathing has attracted considerable research attention and has the potential to improve data transmission performance and maximize resource usage [11], the current MPTCP technology incurs many challenging issues as a transmission protocol for real-time multimedia streaming services. The first major challenging issue when applying MPTCP to stream multimedia content over multi-homed wireless networks is related to the intrinsic fully-reliable nature of MPTCP. As an extension of TCP, MPTCP provides both fully reliable data transmission and strict in-order data delivery, which may significantly degrade the user-perceived quality of service (QoS) of multimedia applications because a single missing data chunk can block other data chunks buffered in the receive buffer from handing over to upper layer [12]. Multipathing with the partial reliability extension has emerged as a preferred solution to satisfy the specific delay requirements for multimedia

applications. However, the current sender-controlled partial reliability extensions [13]–[22] for MPTCP cannot provide a desired QoS guarantees to the time-sensitive loss-tolerant multimedia content delivery, as discussed in the next section.

Another important challenging issue of the MPTCP-based multimedia streaming over a multi-homed wireless network environment is that splitting a multimedia data flow across multiple paths with disparate networking parameters may cause the “hot-potato” packet reordering and receive buffer blocking problems. As previously described, MPTCP allows a TCP connection to utilize multiple access links to aggregate bandwidth and maximize network utilization. However, heterogeneous paths within a MPTCP session can experience different delay performance and will inevitably result in out-of-sequence data arrival at the receiver side [23]. An enormous amount of out-of-sequence data buffered in the “overcrowding” receive buffer can cause serious receive buffer blocking and thus degrades the application performance to a great extent [24]. The most common solution to reduce the out-of-sequence packet arrival problem in MPTCP is to allocate the MPTCP packets to multiple paths based on per-path's RTT performance. However, more and more Internet studies and measurement reports [25]–[27] point out that compared with RTT, one-way delay would be a better choice for the delay-sensitive multimedia applications.

This paper proposes a novel receiver-assisted partial reliable multipath transport solution dubbed as *recPR-MPTCP* for efficient multimedia distribution. The *recPR-MPTCP* is an extension of MPTCP which runs the partial reliability operations at receiver side in order to (i) provide a partially reliable transmission scheme to MPTCP, and (ii) abandon a certain message (e.g., a message whose lifetime is expired) at receiver rather than waiting for the decision from the sender. In addition, inspired by the fact that a receiver can be more clearly and deeply conscious of its adjacent wireless link condition than the sender, the *recPR-MPTCP* also provides a new receiver-based bandwidth aggregation method in which the receiver uses the first-hand one-way delay (forward path delay) to select a set of suitable path for multipathing (*a.k.a.* “partial-MPTCP” mode). More specifically, our proposal has made the following major contributions:

- It presents a receiver-based partial reliability option for MPTCP to abandon expecting late messages, allow the application to go ahead smoothly, and thereby improve end-user experience, by making use of the receiver's intelligence.
- It provides MPTCP with a receiver-assisted multipath management scheme, in which the receiver measures one-way forward delay for each path to help its corresponding sender intelligently aggregate bandwidth for multimedia multipathing.

II. RELATED WORK

In the last few years, the ever-increasing research interest in the MPTCP technology [7] has resulted in considerable

publications. We here categorize the most relevant literature into two groups: **MPTCP scheduler optimization cases** and **MPTCP-based multimedia transmission cases**.

A. MPTCP SCHEDULER OPTIMIZATION CASES

Most of the existing research on MPTCP focuses on the scheduler optimization. Xue *et al.* [28] took into consideration the advantages of forward prediction mechanism and designed a dynamic packet scheduling algorithm to guarantee data delivery performance in MPTCP-based multipath transmissions. Kimura *et al.* [17] investigated the impact of packet allocation strategies on the throughput performance of MPTCP and proposed three alternative schedulers for MPTCP, which are highest sending rate based scheduler, smallest time base scheduler, and largest congestion window (cwnd) size based scheduler. Dong *et al.* [29] developed a loss-aware packet scheduling algorithm to optimize the performance of MPTCP in a highly lossy network environment. Zhang *et al.* [30] designed a novel online subflow detection and association algorithm for MPTCP by jointly considering both the statistical characteristics and protocol information of a data sequence number.

An increasing number of researchers introduced the network coding technologies and cross-layer activities to optimize the MPTCP packet scheduling. Xu *et al.* [14] applied the pipeline network coding technology to MPTCP (dubbed MPTCP-PNC) to optimize the goodput performance of MPTCP packet transmission. They also introduced a quality-based packet scheduling algorithm to enhance the performance of MPTCP-PNC. Cui *et al.* [31] considered the advantages of the intrinsic random characteristic of the fountain code technology and proposed a novel fountain code-based MPTCP packet allocation algorithm. Xue *et al.* [18] first discussed the unfair congestion control problem when applying network coding technology to MPTCP, then they proposed a novel end-to-end congestion control mechanism to resolve the unfairness issue among MPTCP subflows. Lim *et al.* [32] designed a “MAC-MPTCP” cross-layer path management mechanism, which uses MAC layer parameters to calculate the quality and connectivity of MPTCP paths. Fukuyama *et al.* [33] presented a cross-layer packet loss detection mechanism that monitors the frame error in data-link layer to improve MPTCP’s transmission efficiency.

Lately, many researchers have been devoted to optimizing the fairness issues when applying MPTCP for packet delivery. Ferlin *et al.* [34] introduced MPTCP a shared bottleneck detection method in order to make it remain fair to the TCP traffic flows. Zhao *et al.* [35] a fluid-based MPTCP algorithm to necessitate the following aims: (i) making MPTCP be fair to the competing TCP traffic, (ii) achieving load sharing and balancing congestion on multiple paths, and (iii) improving the MPTCP throughput/goodput performance. Sridharan *et al.* [36] designed centralized and distributed heuristic algorithms to judiciously enable

multiple connections at the cell edge to boot fairness in MPTCP-enabled cellular networks.

Apart from the fairness-related problems, there are also energy consumption-related research aspects. Kaup *et al.* [37] investigated the energy consumption of MPTCP. In particular, their research aimed to find the answers of the following three questions: (i) What is the energy consumption of CBR streaming when running MPTCP on today’s mobile devices? (ii) Can the concurrent use of multiple network interfaces decrease the battery power consumption? And (iii) What is the best way to save power when applying MPTCP to a power-constrained device? Wu *et al.* [13] designed an energy usage analytical framework and an Utility Maximization Theory (UMT)-based video content scheduling algorithm to decrease energy cost while still remaining the user-perceived quality of video streaming services. Zhao *et al.* [38] proposed an energy-saving solution for MPTCP-based datacenters by using a flow-completion-time minimized congestion control algorithm and an extra subflow elimination mechanism.

B. MPTCP-BASED MULTIMEDIA TRANSMISSION CASES

More recently, applying MPTCP for real-time multimedia distribution has aroused great research interests. Wu *et al.* [39] presented a novel variant of the MPTCP which combines the video rate allocation and forward-error-correction coding technologies. Moreover, they designed a multipath video transmission-oriented analytical framework to model the performance of MPTCP-based mobile video multipathing. Ferlin *et al.* [40] incorporated the forward error correction into MPTCP to improve the performance of wireless video streaming. Wu *et al.* [41] introduced a variant of MPTCP that aims to balance a trade-off between delay and energy in multipath transmission while guaranteeing the video delivery quality in wireless communication environments, by using a quality-energy tradeoff analytical framework and an energy efficient subflow allocation algorithm.

Other researchers have concentrated their efforts on designing partial reliability extensions to MPTCP. Xu *et al.* [42] proposed a partial reliable MPTCP extension (*a.k.a.* the initiative PR-MPTCP) that allows both MPTCP sender side and receiver side support partially reliable data transmission operations for the delay-constrained applications. Qin *et al.* [43] presented an enhanced partial reliability extension to MPTCP based on the initiative PR-MPTCP. The enhanced partial reliability extension allows the sender to give up the transmission of an invalid chunk by negotiating with its corresponding receiver. Moreover, the enhanced partial reliability extension also included a message-oriented retransmission strategy in order to improve the discarding efficiency. Diop *et al.* [44] introduced the “partial reliability” concept to MPTCP and proposed a new QoS-oriented partial reliability extension to MPTCP for efficient interactive video applications distribution over multiple paths. Our previous work PR-MPTCP⁺ [45] presented a context-aware Quality of Experience (QoE)-oriented MPTCP variant to support partial-reliable multimedia multipathing.

However, to the best of our knowledge, the existing partial reliable MPTCP extensions (i.e., the initiative PR-MPTCP [42], the Message-Oriented Partial-Reliability MPTCP [43], the QoS-oriented MPTCP [44], and the PR-MPTCP⁺ [45]) are the traditional sender-centric solutions, in which the sender performs all key tasks (e.g., flow/congestion control, path quality estimation), while the receiver is limited to only provide feedback information by using the acknowledgment chunks (a.k.a. “ACK chunks”). In contrast, the *recPR-MPTCP* solution includes an “inter-node collaborative transmission” concept, that is, the *recPR-MPTCP* receiver not only participates in and contributes to multipath transmission by giving feedback information to the sender, but also acts as a decision-maker to make meaningful decision according to its obtained first-hand knowledge. For convenience, we briefly introduce the basic mathematical notations and acronyms used throughout this paper, as shown in Table 1.

TABLE 1. Basic notations.

Acronym	Definition
DSN	data sequence number
TSN	transmission sequence number
SACK	selective acknowledgment
RTT, OWD	round-trip time, one-way delay
FD, RD	forward delay, reverse delay
Symbol	Definition
$RTT_{basicSample}$	a basic RTT sample
$FD_{one-way}$	one-way forward delay
$RD_{one-way}$	one-way reverse delay
p_{list}^A	the active path collection
p_{list}^U	the underperforming path collection
p_ℓ^A	the ℓ^{th} path within the p_{list}^A
p_τ^U	the τ^{th} path within the p_{list}^U
Ω	the number of paths in p_{list}^A
$FD_{p_\ell^A}$	the one-way FD value of p_ℓ^A
$FD_{p_\tau^U}$	the one-way FD value of p_τ^U

III. PROBLEM STATEMENT

For the delay-sensitive loss-tolerant characteristics of multimedia applications, using the fully-reliable MPTCP to provide completely reliable data transmission may bring unfavorable influence to user-perceived quality of services. Taking Fig. 2 as an example, when the MPTCP sender allocates packets over multiple asymmetric paths, a multimedia data with a lower DSN (DSN_i) may require a longer time to arrive at the receiver side than a mul-

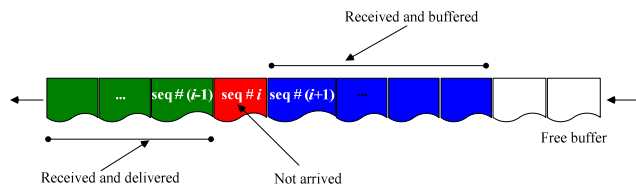


FIGURE 2. An MPTCP receiver buffer contains a missing multimedia data: an example.

timedia data with a higher DSN (DSN_{i+1}). Since the MPTCP technology is designed to guarantee completely reliable data delivery, therefore, the data DSN_{i+1} cannot be flushed to upper layer until the data DSN_i is received by the receiver successfully, regardless of whether the playing deadline of DSN_{i+1} is come or not. Such fully-reliable transmission mechanism will undoubtedly degrade the user-perceived QoS of multimedia applications.

Multipathing with partial reliability operations is considered to be the most beneficial to enhance the user-perceived QoS, by giving up on an expired or a relatively unimportant multimedia data chunk and flushing the data chunks whose playing time are coming up to upper layer timely. Unfortunately, although there has been a growing recent interest in applying the “partial reliability” concept to MPTCP, there is yet no standardized partial reliability extension for the MPTCP to support real-time multimedia communications and enhance user’s multimedia perception. Moreover, the current existing partially reliable MPTCP solutions [42]–[45] are more or less following the design ideas of the standardized PR-SCTP (a partially reliable extension of Stream Control Transmission Protocol) [46]. In order to clarify the problems further, below we briefly introduce how PR-SCTP enables the partially reliable transmission service for real-time multimedia applications:

- 1) During the period of association setup, the sender and the receiver need to negotiate the partially reliability extension, as shown in Fig. 3(a);
- 2) During data transmission, the sender will give up on retransmitting a missing multimedia data when its lifetime is expired, advance the expected TSN value, and notify the receiver by sending a Forward-TSN chunk ($FWD\ TSN\ 3$);
- 3) When the Forward-TSN chunk ($FWD\ TSN\ 3$) is received, the receiver makes available all received multimedia data up to the new point and then continues on, as shown in Fig. 3 (b).

Through the above partially reliability operations, PR-SCTP can provide a flexible balance between reliabil-

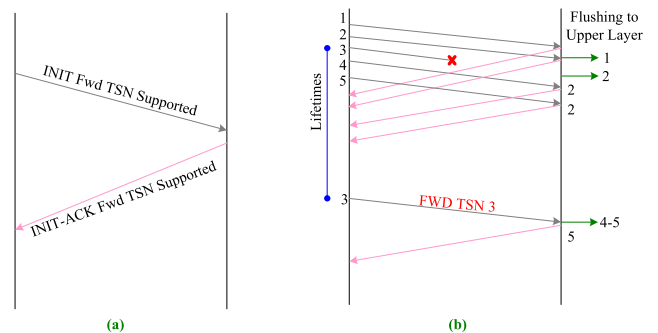


FIGURE 3. Partially reliability operations in SCTP: (a) Partially reliability extension negotiation during association setup, and (b) Forward an expired multimedia data with TSN 3.

ity and timeliness for real-time multimedia data transfer. However, there are still some challenges in PR-SCTP to be addressed. The first challenge is that it would take a long time for a Forward-TSN chunk arrival at receiver from its corresponding sender. The late arriving Forward-TSN chunk (e.g., FWD TSN 3 in Fig. 3(b)) may cause that some multimedia packets in receiver buffer (e.g., packets with TSN 4 and 5 in Fig. 3(b)) cannot be flushed to upper layer before their playing time. The second challenge is also related to the Forward-TSN chunk. In practice, a Forward-TSN chunk can be dropped and/or lost for any number of different reasons at different intermediate nodes along the network paths from the sender to the receiver. As Fig. 4 shows, the FWD TSN 5 is accidentally dropped and/or lost in the process of transmission. Due to the missing FWD TSN 5 chunk, the receiver cannot timely hand over the packets with TSN 6, 7 and 8 to upper layer, regardless of the playing time of those packets is coming up.

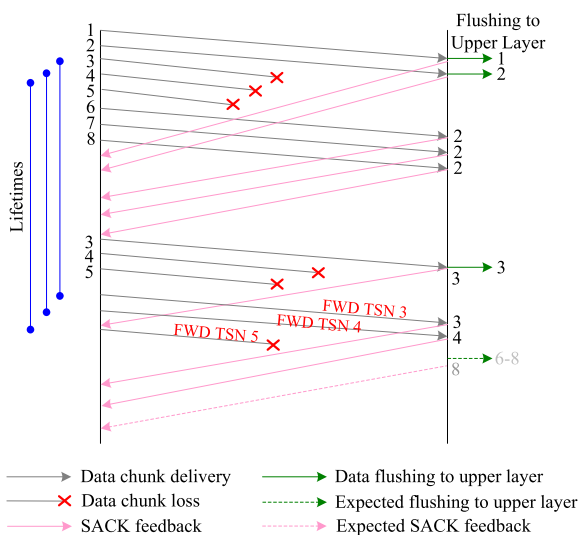


FIGURE 4. A Forward-TSN chunk (FWD TSN 5) is dropped and/or lost in the process of transmission.

The above deficiencies in PR-SCTP also exist in the current PR-SCTP-like partially reliable MPTCP solutions, and are unfavorable for the real-time multimedia streaming services. Taking Fig. 3(b) for example, we assume that the packet with TSN 3 is lost and its lifetime is expired, the sender decides to abandon the expired message and initiates a Forward-TSN chunk (FWD TSN 3 in Fig. 3(b)) to tell the receiver to flush the buffered data (with TSN 4 and 5 in Fig. 3(b)) to the application layer. For this sender-centric case, the time required for the receiver to know the message abandonment notification and then go ahead, denoted as $t_{msg_flushing}$, can be expressed by

$$t_{msg_flushing} = t_{sender_proc} + t_{FWD_DSN} + t_{rec_proc}, \quad (1)$$

where t_{sender_proc} , t_{FWD_DSN} and t_{rec_proc} are the time the sender takes to initiate a Forward-TSN chunk (e.g. FWD TSN 3), the time the Forward-TSN chunk takes to travel

from the sender side to the receiver side, and the time needed for processing the Forward-TSN chunk at the receiver side, respectively. In contrast, if the receiver does not need to wait for the sender’s notification, instead, it directly makes a decision to flush these packets to upper layer for playing, then the time overhead ($t_{msg_flushing}$) can be reduced to a minimum, which can be expressed by

$$t_{msg_flushing} = t_{rec_proc} . \quad (2)$$

Moreover, out-of-sequence packet receipt is a common and unavoidable phenomenon in a multi-homed wireless network environment because heterogeneous paths with different delay characteristics are likely to become more common. The out-of-sequence arrival phenomenon is obviously problematic in MPTCP since numerous out-of-sequence data maintained in the space-limited receive buffer will undoubtedly cause receive buffer blocking and lead to severe performance degradation in multipath transmission.

In recent years, a growing interest in the MPTCP out-of-sequence problem has resulted in many useful solutions [23], [24]. Among these solutions, allocating application data over multiple paths according to their own RTT values may seem to be a very simple and straightforward approach to mitigate the out-of-sequence data reception problem. However, these RTT-based solutions are designed with a basic assumption that within each RTT, both the FD and RD are similarly equal to each other (namely each of FD and RD is roughly $\frac{1}{2} \times RTT$). In fact, a number of Internet traffic measurements reveal that the delays are typically asymmetric along the forward and reverse directions of a network path due to asymmetric routing (a.k.a. network path asymmetry), as shown in Fig. 5.

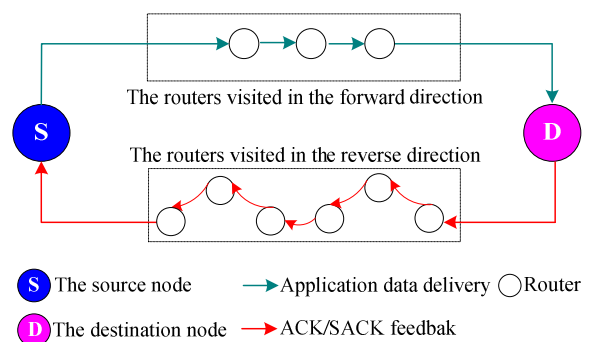


FIGURE 5. The routers visited in the forward direction (from S to D) and reverse direction (from S to D).

The asymmetric routing can be especially problematic in MPTCP. Taking Fig. 6 as an example, although all the forward and reverse delays of path A and path B are different, the sum of forward delay and backward delay of the two paths (namely RTT) is equal. Correspondingly, the two paths may be considered to have the same transmission delay performance, which would serious influence the decision of the MPTCP sender. Since RTT often fails to identify specific QoS issues in the intrinsic asymmetric routing nature of

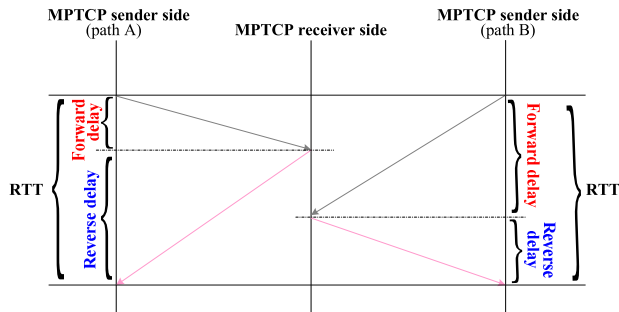


FIGURE 6. The delays are typically asymmetric along the forward and reverse directions of a network path.

wired/wireless network environment, it would benefit more from using one-way delay for MPTCP to enhance the performance and QoS of data transmission. Recent research has demonstrated that applying forward delay to MPTCP for packet scheduling over multiple paths is of great potential in out-of-sequence data arrival mitigation [47]. However, the potential of the OWD metrics, especially combining with the receiver's intelligence, has not been taken into account in the existing MPTCP path management mechanisms.

Considering all the above problems, this paper presents a receiver-assisted partial reliable multipath transport solution for real-time multimedia applications, which is called as *recPR-MPTCP*. The main objectives of *recPR-MPTCP* are (i) to overcome the two shortcomings of the MPTCP mentioned above, and (ii) to satisfy the specific QoS requirements for multimedia applications while maximizing network resource utilization efficiency. It is particularly worth mentioning here that we in this paper focus exclusively on MPTCP, the concept of a receiver-assisted partial reliability could be also an interesting proposal for the SCTP and other multi-homing protocols.

IV. RECPR-MPTCP DETAIL DESIGN

Fig. 7 illustrates the system framework of the designed *recPR-MPTCP*, which consists of two MPTCP endpoint (a sender and its corresponding receiver), and n multiple paths. The *recPR-MPTCP* sender, like that of MPTCP, divides the application data into several segments, allocates the traffic over the n available paths, and runs the flow/congestion control mechanisms. The *recPR-MPTCP* receiver, also like that of MPTCP, collects and reorders these data sent from the corresponding sender. Apart from the above operations inherited from the standard MPTCP, there are two new modules, which are *stream partial-reliability engine* (SPE) and *stream multipathing engine* (SME). The main functions of both SME and SPE are introduced as follows:

- The SPE runs at the receiver side, it is devoted to enabling the partially reliability operations, measuring the one-way FD values for each path, and assisting the sender to run the partial MPTCP operations.
- The SME runs at the sender side, it is dedicated to adaptively aggregating bandwidth of multiple network

links for effective data delivery according to the explicit instruction (SACKs) from the receiver.

A. STREAM PARTIAL-RELIABILITY ENGINE

In the current MPTCP partially reliability solutions, the receiver plays a passive role in the relationship. For example, the receiver cannot drop any packet without explicit instruction from the corresponding sender, it cannot distinguish a late arriving packet, it also cannot notify the sender about the late arriving packet. Such sender-centric receiver-passive solutions cannot make multimedia users achieve very high satisfaction, like it was previously analyzed. In contrast, implementing the partially reliability functions on the receiver side has its potential advantages in the real-time multimedia delivery mainly due to (i) the receiver can be capable of exploiting the first-hand information obtained on the forward path for partial reliability operations, instead of giving feedback and then waiting for the decisions sent from the sender, (ii) the receiver should know better than the sender about the transmission characteristics of wireless channels, as well as the status information of receive buffer.

By considering the above reasons, we implement the SPE module on the *recPR-MPTCP* receiver side for efficient multimedia delivery. The main objectives of the SPE module are (i) to provide the upper layer with a partial reliable data mode in order to guarantee a desired frame rate and ensure the QoS of real-time multimedia applications, and (ii) to provide the sender with the path selection assistance in order to alleviate the packet reordering and receiver buffer blocking problems. To achieve the objective (i), for a missing segment who is required to be resent but its lifetime is going to be expired the SPE takes the following actions to provide partial reliable services: (a) neglecting this missing segment then moving the cumulative ACK point forward, and (b) notifying its corresponding sender (with the SACK chunk) to abandon that retransmission of the data.

Another reason for the SPE to abandon a missing multimedia data is related to the receiver buffer blocking. When severe receiver buffer blocking occurs, the SPE will neglect a missing multimedia data according to its reliability level (importance) in order to possibly free the receiver buffer. For example, at the multimedia frame level, an I-frame is common with the highest level of reliability than other frames (a P- or B-frame). Therefore, in the case of severe receiver buffer blocking, the SPE will neglect the missing B or even P-frames and move the cumulative ACK point forward, then notify sender to abandon that retransmission of the frames. This is a necessary way for multimedia applications because receiver buffer blocking may prevent the sender from sending any new segment to the receiver until the receive buffer space is enough again for the new arrival segments.

Apart from the message abandonment due to expiration and reliability level, the SPE can also abandon a missing data that blocks the other data from handing over to the upper layer. As discussed in Section II, out-of-sequence arrival is an inevitable phenomenon when data arrives at the

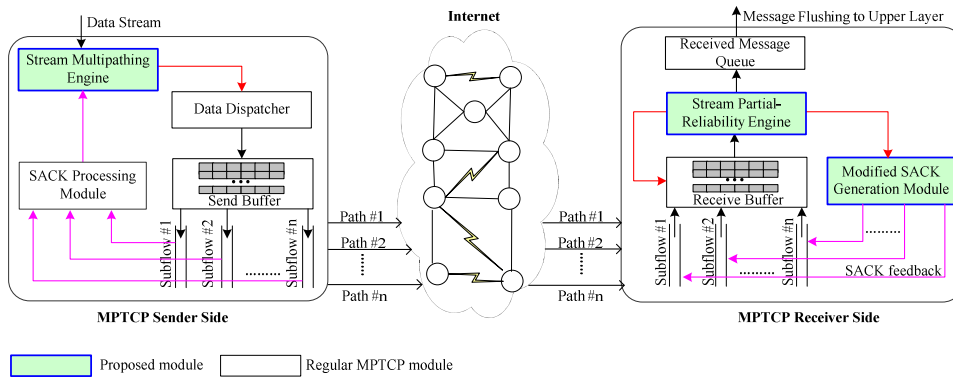


FIGURE 7. The architecture of recPR-MPTCP.

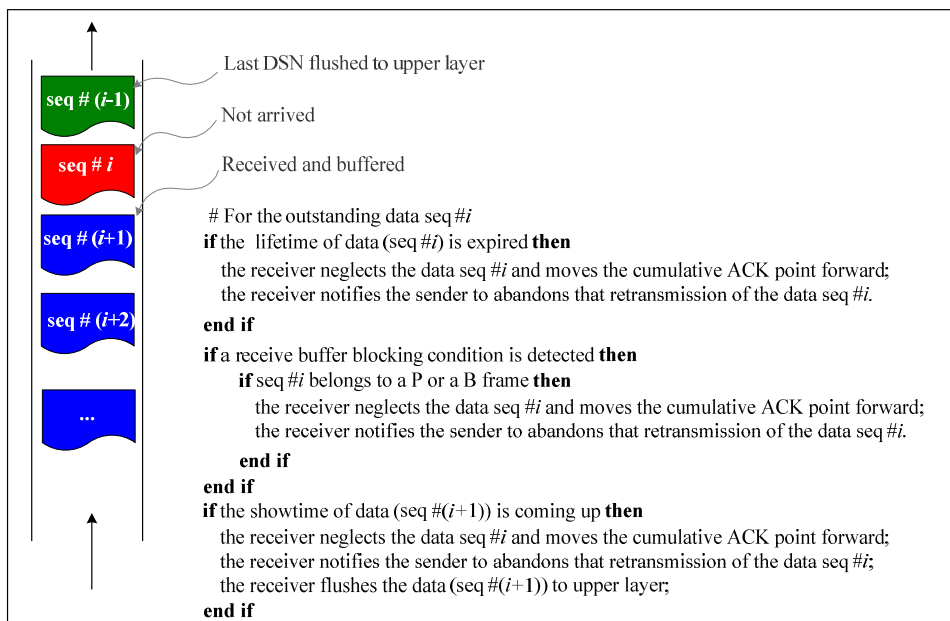


FIGURE 8. SPE-based Message abandonment and handover algorithm.

receiver over multiple network paths. Meaning that during the MPTCP-based multimedia transmission session, a data with a higher DSN (DSN_{i+1}) may have already arrived in the receiver buffer and awaiting the arrival of a data with a lower DSN (DSN_i), even if its lifetime is going to run out. For the case when the playing time of DSN_{i+1} is coming while DSN_i is still absent, the SPE will flush the data DSN_{i+1} to the upper layer, move the cumulative ACK point forward, and notify the sender to abandon that retransmission of the data DSN_i . The SPE-based message abandonment and handover algorithm (dubbed as Algorithm 1) is clearly illustrated in Fig. 8.

To achieve the objective (ii), the SPE needs to measure each MPTCP paths' one-way FD value, which is increasingly used by the Internet Service Providers (ISPs) to ensure the QoS of real-time applications [48], to assist the corresponding sender to adaptively use a subset of suitable paths for multipathing. Suppose there are n paths (p_1, p_2, \dots, p_n)

within an MPTCP session and taking an available path p_ψ ($p_\psi \in (p_1, p_2, \dots, p_n)$) for example, the SPE calculates an initial FD sample of path p_ψ , t_1 , by using the formula from Eq. (3),

$$t_1 = t_{rec} - t_{send} - \Delta t, \quad (3)$$

where t_{send} is the time stamp of a packet when it was sent by an MPTCP sender, t_{rec} is the time stamp of the packet when it arrived at the receiver side, and Δt is the packet processing time (this value can be ignored during the sample calculation).

After collecting the FD sample as presented in equation (3), we calculate the average FD value for each path (in order to avoid bursty fluctuation). Let us suppose that the values of the FD samples in p_ψ are t_1, t_2, \dots, t_m , the SPE calculates

the average value of the FD samples by,

$$\overline{T}_M = \frac{1}{M} \times \sum_{i=1}^M t_i, \quad (4)$$

where t_i ($t_i \in (t_1, t_2, \dots, t_m)$) is a value of the FD sample in path p_ψ , M and \overline{T}_M are the total number of the FD samples and the average value of the FD samples, respectively.

Eq. (4) is a general expression used to calculate the average FD value. In order to avoid maintaining all the FD samples at the receiver side, the SPE uses an iterative method (see below equation (5)) to compute the average FD value for path p_ψ ,

$$\overline{T}_{M+1} = \frac{1}{M+1} \times (\overline{T}_M \times M + t_{M+1}), \quad (5)$$

By using the average FD value of each path, the SPE can sort all the MPTCP paths in a descending order. When the receiver suffers from buffer blocking, the SPE gives the sender feedback on an underperforming path (the first path within the sorted path group) to assist the sender in aggregating bandwidth for data multipath transmission, as we will detail later.

B. STREAM MULTIPATHING ENGINE

Although MPTCP has tremendous potential of becoming a core transport layer technology for data delivery, the full-MPTCP mode-based path management mechanism in RFC6824 (where all available paths are used for data transmission) is too simple to effectively distinguish path transmission conditions. When the sender assigns traffic over all activated paths and performs the congestion control algorithms separately for each path, each path has its own *cwnd* size and is independently responsible for data delivery. Nevertheless, different paths are common with different delay characteristics and sensitive to variations in wireless transmission condition. These paths, as previously stated, can influence each other interactively since MPTCP provides the same reliable and in-order delivery of data to the application as the traditional single-path TCP.

To make the best use of multiple paths with MPTCP and alleviate the out-of-sequence packet arrival problem due to delay difference, a path with a huge delay compared to other paths should be deactivated for data delivery, as shown in Fig. 9. In other words, the sender “turns on” a partial-MPTCP transmission mode to prevent the usage of paths with large delay in multipathing, and thus to alleviate buffer blocking at receiver side. The current RTT-based MPTCP path management mechanisms mostly consider the RTT, which mainly consists of the forward and reverse delays (see Eq. (6)), as performance metric to manage multiple paths,

$$RTT_{basicSample} = FD_{one-way} + RD_{one-way} + \Delta t, \quad (6)$$

where Δt is the packet processing time.

From Eq. (6), it can be seen that as the sum of a forward delay and a reverse delay, RTT often fails to fully reflect the

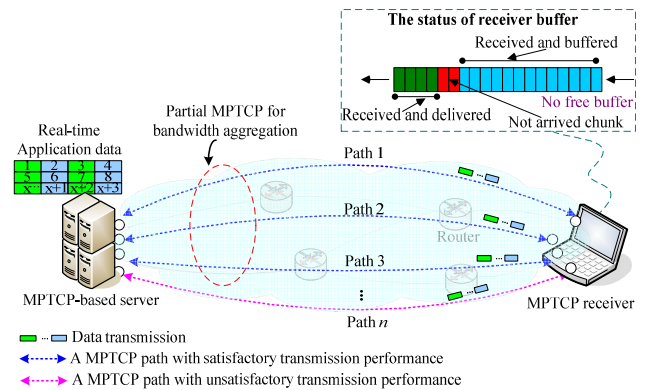


FIGURE 9. Partial MPTCP for bandwidth aggregation and parallel transmission.

transmission quality characteristics because the delays along the forward and reverse directions of a network path are often different. If the MPTCP uses RTT as metric to to disable or enable a path in multipath transmission, a path with a satisfactory forward delay and a relatively large reverse delay may be deactivated for data delivery, while a path with an unsatisfactory forward delay but with a relatively small reverse delay may be retained use in MPTCP. To address this issue, the SME module, which is running on the *recPR*-MPTCP sender side, uses a one-way FD-based multipath management scheme to dynamically select a subset of suitable paths for multipath data transmission, assisted by the receiver-based SPE module.

The main workflow of collaborative process between SME and SPE is described briefly below: (i) the SPE module calculates per-path’s one-way FD values once an application data is received successfully, (ii) the SPE module sorts MPTCP paths in a descending order based on the measured one-way FD values of these path, (iii) the SPE module monitors the status of receiver buffer, when the receiver buffer blocking occurs, it advertises the first path information (the path with the largest one-way FD characteristics) to the SME module by using a SACK chunk, (iv) when the SACK chunk including the path advertisement notification arrives, the SME module marks the path as “Underperforming” and prevents it from sending data. Moreover, in order to ensure that *recPR*-MPTCP does not perform worse than the standard MPTCP, an underperforming path can be marked as “Active” and reused for data delivery. For example, an underperforming path p_τ^U can be put into p_{list}^A and reactivated for data transmission when the below condition is satisfied,

$$FD_{p_\tau^U} - \frac{1}{\Omega} \times \sum_{\ell=1}^{\Omega} FD_{p_\ell^A} \geq 0. \quad (7)$$

Through collaboration works with the SPE module, the SME can effectively manage multiple paths and adaptively transform the transmission model from full-MPTCP mode to partial-MPTCP model (or vice versa) for bandwidth aggregation and parallel transmission. Such cooperation

way also has the potential to ensure packets in-order arriving, avoid unnecessary packet reordering delay, mitigate receiver buffer blocking, and increase application-level performance. The pseudo-code of main procedures of SPE-SME cooperation-based multipathing is presented in Algorithm 2.

Algorithm 2 SPE-SME Cooperation-Based Multipathing Algorithm

```

/*Once having data to be sent, SME takes the following actions*/
1: for ( $\ell = 1, \ell \leq \text{count}(p_{list}^A), \ell ++$ ) do
2:   assign data to the path  $p_{\ell}^A$ ;
3: end for
/*When a data is received, SPE takes the following actions*/
4: for ( $\ell = 1, \ell \leq \text{count}(p_{list}^A), \ell ++$ ) do
5:   calculate one-way FD value for  $p_{\ell}^A$ ;
6: end for
7: sort all the paths in  $p_{list}^A$  according to their own FD value;
8: send the path with the largest FD value to the sender when the receiver buffer blocking occurs;
9: for ( $\tau = 1, \tau \leq \text{count}(p_{list}^U), \tau ++$ ) do
10:  calculate one-way FD value for  $p_{\tau}^U$ ;
11:  if  $FD_{p_{\tau}^U} - \frac{1}{\Omega} \times \sum_{\ell=1}^{\Omega} FD_{p_{\ell}^A} \geq 0$  then
12:    advertise the sender to put  $p_{\tau}^U$  into  $p_{list}^A$  by SACK;
13:  end if
14: end for
    
```

V. PERFORMANCE TESTING

A. SIMULATION SETUP

In the experiment, the Network Simulator version 2 (more commonly known as NS-2) [49] with the standard MPTCP patch [50] has been used to study the performance of our proposal. Fig. 10 illustrates the simulation topology that involves two MPTCP endpoints who are connected through three paths (A, B, and C) simultaneously. In order to fully investigate the effect of delay variations on the MPTCP behaviors, all the three MPTCP paths are set with the same bandwidth of 11 Mbps and 5-50 ms propagation delay time. The core network bandwidth on the three paths are set to 100Mbps.

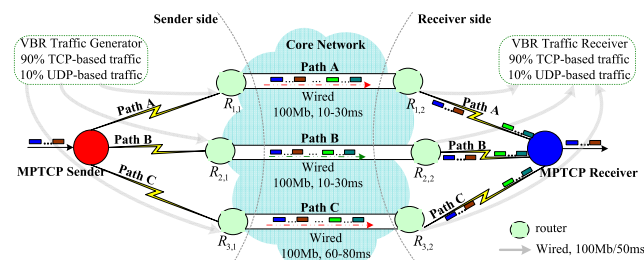


FIGURE 10. Simulation topology.

TABLE 2. Path settings.

Parameters	Paths A & B	Path C
Wireless network bandwidth	11Mbps	11Mbps
Wireless network delay	5-10ms	40-50ms
Queue management mechanism	Drop-tail	Drop-tail
Wired link bandwidth	100Mbps	100Mbps
Wired network delay	10-30ms	60-80ms
Uniform loss rate	1%-2%	5%-10%
Markov loss rate	1%	1%

Table 2 presents the major parameters of the three MPTCP paths.

In addition, we attach two wireless loss models, *Uniform loss model* (to represent distributed frame drop due to wireless interference) and *Gilbert-Elliot model* (to represent abrupt continuous frame loss due to weak signal), for each wireless Wi-Fi link in order to perform the link-layer frame loss. We also attach each path with a VBR traffic generator to emulate the Internet background traffic. The packet size selected for the VBR traffic are as below: 49% are 44 bytes in length, 2.1% are 628 bytes, 2.9% are 576 bytes, 46% are 1500 bytes. The competing VBR traffic on paths A and B consumes 0-50% of the wireless access bandwidth, while on path C, it consumes 80-100% of the wireless access bandwidth (in order to simulate the network congestion caused by burst background traffic injection).

We conduct comprehensive performance evaluation of *recPR-MPTCP* under the conventional reliable FTP traffic and real-time video traffic, respectively. In the simulation of FTP traffic transmission, the *recPR-MPTCP*'s SME module is enabled only in order to investigate how *recPR-MPTCP* can benefit from the SME module. To better illustrate the comparison, we only compare our solution with the baseline MPTCP [7]. It should be noted that the existing partial-reliable MPTCP variants are not involved in the comparison. This is because that these partial-reliable MPTCP variants mostly inherit the basic operations from the baseline MPTCP, such as congestion control, packet scheduling and so on. In other words, the existing partial-reliable MPTCP variants can possibly perform same behaviors as the baseline MPTCP in the transmission of traditional FTP-like data traffic when their partial-reliability options are disabled.

In order to investigate the performance of *recPR-MPTCP* in real-time video transmissions, we use the Evalvid toolkit [51] together with the NS-2 interface code to evaluate quality of a transmitted video. A YUV video test sequence consisting of 2000 frames (a QCIF format with 176 pixels wide and 144 pixels tall) is chosen as an original video sequence [52]. To emulate massive video flows and achieve meaningful results, we concatenate the YUV video test sequence to be 10000 frame-long (by concatenating five of the original video sequences into one single video sequence). The structure of Group of Pictures (GoP) is IPBBPBB,

the end-to-end transmission delay constraint specified for each GoP is set equal to the showing duration (e.g., 500 ms). After the preprocessing procedure, an MPEG-4 video test sequence is generated with 1115 Intra frames (commonly known as I-frame), 2225 Predictive frames (commonly known as P-frame), and 6660 Bi-predictive frames (commonly known as B-frame). The 1000 frames are divided into 11250 packets, in which 6670 packets for B frames, 2265 packets for P frames, and 2315 packets for I frames. The corresponding packet information is fed to NS-2 via the generated MPEG-4 video trace file. These 11250 packets will be transmitted over the MPTCP paths.

In the simulation of FTP data transmission, the simulation time is set to 100 seconds with infinite FTP flows. In the simulation of video traffic transmission, the MPTCP sender transfers the testing video trace file to the MPTCP receiver, and the simulation will stop after the sender finishes the transmission of all the video frames. All the experimental results are calculated by averaging the testing values of the 20 runs.

B. FTP DATA TRANSMISSION COMPARISON

1) DATA SENDING AND RECEIVING TIMES

Fig. 11 presents the data sending and receiving times (DSRT) when the two technologies are adopted, respectively. We can see that *recPR-MPTCP* achieves a higher level of DSRT than the baseline MPTCP. The reason is that the baseline MPTCP uses all the available paths for data transmission and fails to consider that an unsatisfactory path that has large forward path delay can constrain the data transmission performance, by restricting the sender to assign new packet to the stable paths. Particularly, the baseline MPTCP scheduler can also cause receiver buffer blocking, with a great amount of out-of-sequence data buffered in the “overcrowding” receiver buffer. By contrast, *recPR-MPTCP* can prevent the usage of unsatisfactory paths, adaptively aggregate bandwidth of satisfactory paths for multipathing, thereby it attains a higher level of DSRT than the baseline MPTCP.

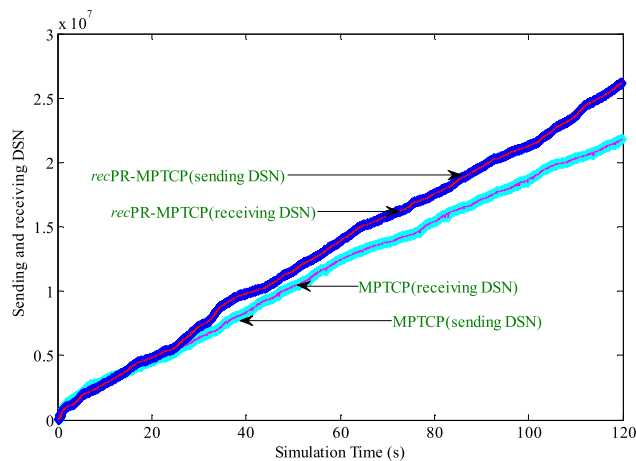


FIGURE 11. The comparison of Data sending and receiving times.

2) OUT-OF-SEQUENCE DSN

The out-of-sequence DSN is considered as a suitable performance metric for the evaluation of multipath protocols. The out-of-sequence DSN is calculated by the sequence number offset between two consecutively segments arrived at the receiver side. Fig. 12 illustrates the out-of-sequence DSN comparison when the two technologies are adopted, respectively. We can see that the proposed *recPR-MPTCP* generates less out-of-sequence DSN than the baseline MPTCP. The reason is because the baseline MPTCP transfers the traffic data using all the available paths in the connection, without considering that a huge delay difference among these paths will lead to an extremely large amount of out-of-sequence data for reordering. By contrast, *recPR-MPTCP* disables an underperforming path in multipath transmission and only selects suitable paths for data delivery. In this way, *recPR-MPTCP* can guarantee the packets to be received by the receiver in the correct order. When comparing the two technologies, it can be observed that the peak out-of-sequence DSN is about 1.4×10^5 when using *recPR-MPTCP*, while it is up to 3.1×10^5 when using the baseline MPTCP.

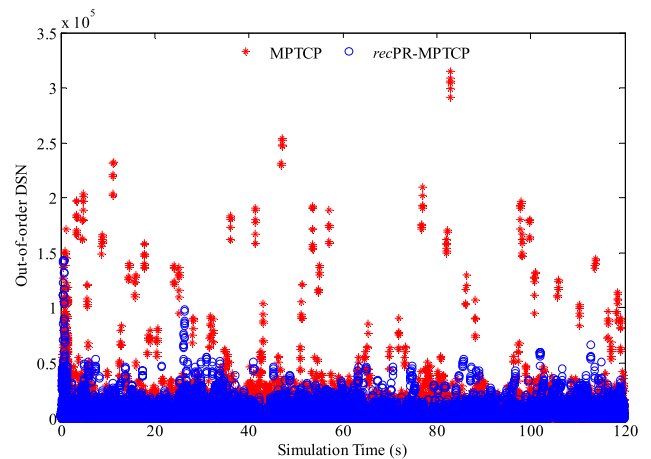


FIGURE 12. The comparison of out-of-sequence DSN.

3) END-TO-END DELAY

Fig. 13 illustrates the results comparison in regard to delay time when using the two schemes, respectively. As mentioned earlier, *recPR-MPTCP* takes into consider the forward delay characteristics during path quality estimation and selection, it distributes packets over these selected paths in proportion to their respective FD values. Therefore, *recPR-MPTCP* can avoid allocating traffic over an underperforming path (with an unsatisfactory forward delay) in multipath transmission, this help *recPR-MPTCP* to reduce the time delay and overhead in both transmission and reordering. Correspondingly, *recPR-MPTCP* achieves lower time delay than the baseline MPTCP. When comparing the overall mean time delay of the two schemes (with the total 120 seconds of the simulation), we can see that the *recPR-MPTCP* solution’s overall mean

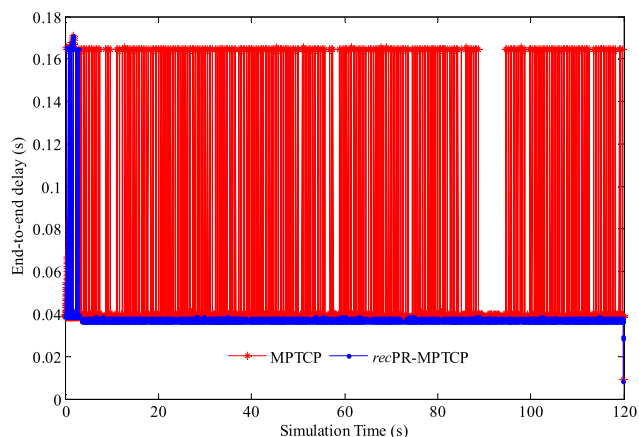


FIGURE 13. Delay comparison.

delay is about 14.91% lower than that of the baseline MPTCP scheme.

4) JITTER MEASUREMENT

In computer networking, jitter can refer to the delay variation in transmitting, propagation and queuing of a packet [53]. Jitter is recognized as a very important metric that is widely used to investigate the performance of a transport technology. A low level of jitter usually happens at a satisfactory transmission protocol. In our experiment, a jitter value is computed by a latency variation between two consecutively segments that are received by the receiver successfully. Fig. 14 illustrates the jitter comparison when the two technologies are used, respectively. As we can see from the figure, the transmission model switching behavior (from full-MPTCP mode to partial-MPTCP model) of the *recPR-MPTCP* has a negative effect on the jitter performance that is sometimes greater than that of the baseline MPTCP, however, *recPR-MPTCP* can detect and differentiate forward path delay in time and allocate traffic based on each path's FD value, it thus can reduce variations in latency (jitter).

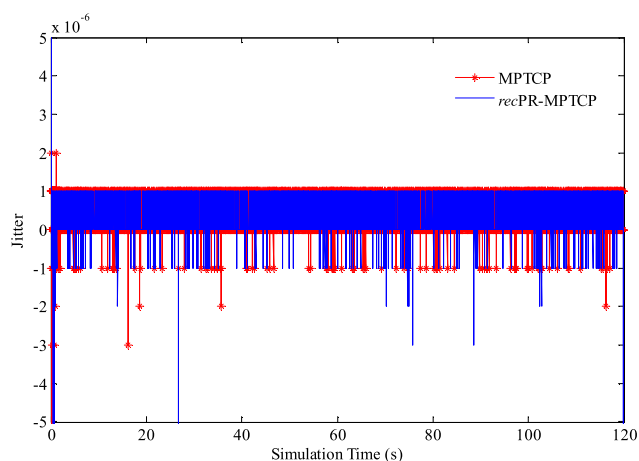


FIGURE 14. Jitter comparison.

5) AVERAGE THROUGHPUT

Fig. 15 illustrates the throughput comparison when using the two technologies, respectively. Since *recPR-MPTCP* includes a new multipath bandwidth aggregation method which uses the one-way FD value obtained by the receiver as each path's transmission quality to select a set of suitable path for multipathing. The one-way FD-based bandwidth aggregation feature possibly prevents *recPR-MPTCP* suffering from receiver buffer blocking, and in turn, enhances the throughput performance of multipath transmission. In contrast, in the baseline MPTCP the packets are received in a different order from which they were transmitted due to the dissimilar forward delay characteristics of multiple paths. Packet arrival in an incorrect order is likely to cause the degradation of the MPTCP throughput performance.

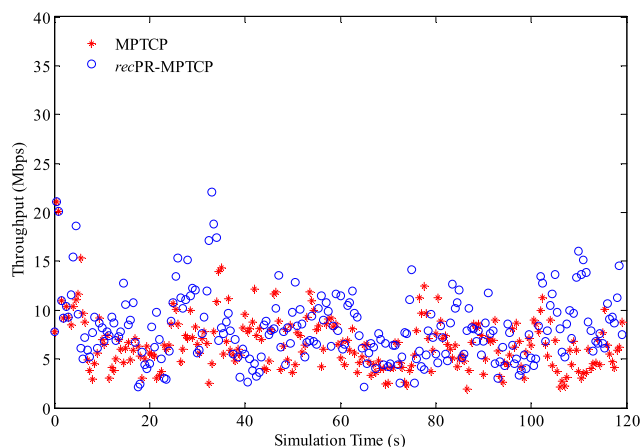


FIGURE 15. Throughput comparison.

In addition to comparison with the baseline MPTCP, we also present the comparison of *recPR-MPTCP* with our previous work $MPTCP+(PU)^2M^2$ - a potentially underperforming-aware path usage management mechanism for MPTCP. Fig. 16 shows the overall mean throughput of the three methods with a total 120 seconds simulation time. Because *recPR-MPTCP* and $MPTCP+(PU)^2M^2$ prevent the usage of underperforming in multipath trans-

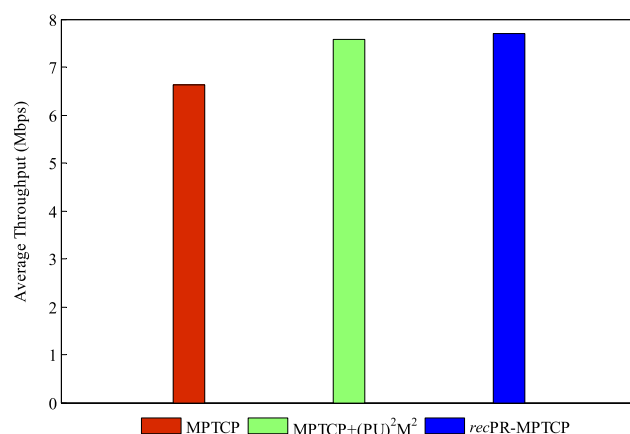


FIGURE 16. Overall mean throughput comparison.

mission accordingly. Therefore, both the two solutions perform better than the baseline MPTCP. When comparing *recPR-MPTCP* and $MPTCP+(PU)^2M^2$, it is noted that the *recPR-MPTCP* achieves higher mean throughput than the $MPTCP+(PU)^2M^2$. This is because that the $MPTCP+(PU)^2M^2$ detects an underperforming or a broken path by monitoring the delay changes of paths. This path estimation method is very sensitive to variations of delay and can cause throughput performance degradation because of frequently changing the paths in wireless transmission. In contrast, *recPR-MPTCP* declares an underperforming path by monitoring the receiver buffer blocking, this way helps *recPR-MPTCP* to provide stable transmission performance. As the figure shows, the *recPR-MPTCP*'s overall mean throughput is 15.17% higher than that of the baseline MPTCP and 1.56% higher than that of $MPTCP+(PU)^2M^2$.

C. REAL-TIME VIDEO DELIVERY COMPARISON

This subsection investigates how *recPR-MPTCP*'s performance compares with that of the baseline MPTCP, the initiative PR-MPTCP, and the Message-Oriented Partial-Reliability MPTCP schemes in the transmission of multimedia content. In this test, both the SME and SPE modules of *recPR-MPTCP* are enabled. To better illustrate, we portray the results of our solution as '*recPR-MPTCP*', while the results of the initiative PR-MPTCP, the Message-Oriented Partial-Reliability MPTCP, and the baseline MPTCP schemes are portrayed as 'PR-MPTCP', 'MO-PR', and 'MPTCP', respectively.

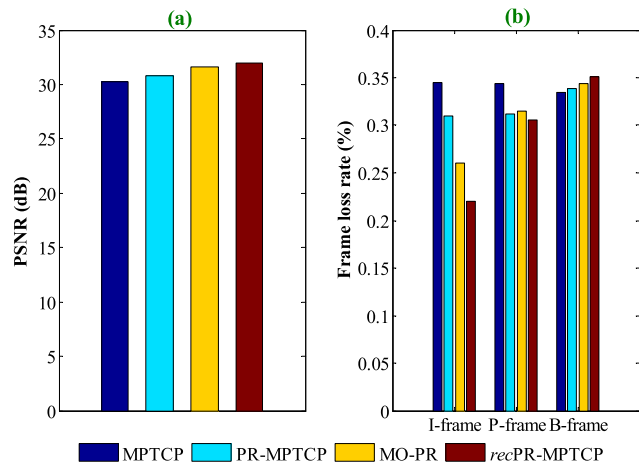


FIGURE 17. The comparisons of (a) average PSNR (db), and (b) frame loss rate.

Figs. 17 (a) and (b) present the comparison results of peak signal-to-noise ratio (PSNR) and frame loss rate when using the baseline MPTCP, initiative PR-MPTCP, Message-Oriented Partial-Reliability MPTCP and *recPR-MPTCP* schemes, respectively. Fig. 17 (a) shows that compared with the other three schemes, *recPR-MPTCP* achieves the highest PSNR. The reason is that *recPR-MPTCP* provides partial reliability operations which allow the

receiver to abandon the invalid multimedia data in multipath transmission. Furthermore, *recPR-MPTCP* differentiates each path's forward delay differences and "turns on" partial-MPTCP transmission model accordingly. The baseline MPTCP uses all the available paths to transfer multimedia traffic, it fails to take into account the forward path delay differences in multipath transmission. Moreover, the baseline MPTCP does not consider the real-time nature of multimedia frames, it may retransmit the expired frames repeatedly, which causes unnecessary delay and leads to performance degradations. As a result, it achieves lower PSNR performance than the other three schemes.

Both the initiative PR-MPTCP and Message-Oriented Partial-Reliability MPTCP schemes can enable a 'partial reliable transmission service' to satisfy the real-time constraints of multimedia data. Correspondingly, the two schemes can attain a higher value of PSNR than the baseline MPTCP. However, both the initiative PR-MPTCP and Message-Oriented Partial-Reliability MPTCP schemes, like the baseline MPTCP, use a conventional and fully multipath transmission way to transfer multimedia traffic, regardless of whether the delay difference among the transmission paths is huge or not. Fortunately, the Message-Oriented Partial-Reliability MPTCP includes a new Message-Oriented retransmission mechanism. This retransmission mechanism helps the Message-Oriented Partial-Reliability MPTCP attain a higher value of PSNR than the initiative PR-MPTCP scheme in highly dynamic network conditions.

TABLE 3. Comparison of VQM and SSIM.

Method	VQM	SSIM
MPTCP	1.569	0.894
PR-MPTCP	1.435	0.895
MO-PR	1.396	0.897
<i>recPR-MPTCP</i>	1.328	0.910

As Fig. 17(a) shows, the mean value of the PSNR difference between *recPR-MPTCP* and the baseline MPTCP is 1.75 dB, it is 1.21 dB between *recPR-MPTCP* and the initiative PR-MPTCP, and it is 0.38 dB between *recPR-MPTCP* and the Message-Oriented Partial-Reliability MPTCP. In addition, we use the MSU Video Quality Measurement Tool (MSU VQMT) [54] to compare the video quality between the original video and the reconstructed video clips. Table 3 shows the performance comparisons among the baseline MPTCP, the initiative PR-MPTCP, the Message-Oriented Partial-Reliability MPTCP and the proposed *recPR-MPTCP* schemes in terms of Structural Similarity (SSIM) and Video Quality Metric (VQM), respectively. As the table shows, *recPR-MPTCP* has the highest value of SSIM and the lowest value of VQM among the four methods compared. The above results indicated that our solution outperforms the baseline MPTCP, the initiative PR-MPTCP and the Message-Oriented Partial-Reliability MPTCP schemes in terms of user-perceived multimedia service quality.

VI. DISCUSSION

From the simulation results, we can see that the throughput performance improvements of the proposed *recPR-MPTCP* is limited compared to our previous work *MPTCP+(PU)²M²* (the average throughput is only increased by 1.56%). However, it is worth mentioning that the *recPR-MPTCP* solution is the first study to run the message abandonment decisions on the receiver side, by using the receiver's own intrinsic advantages and intelligence. The main goals of this paper are: (i) to introduce a new way of considering partial reliable ordering, not limited to the *MPTCP*, but also other transport technologies are seen in a new light, and (ii) to attract more researchers to explore further the most effective uses for partial ordering whether in the promising *MPTCP* technology, or other connection-oriented transport protocols. We here discuss the limitations of our proposal and highlights the open and interesting problems. We hope to attract more researchers pay attention to this interesting topic and drive this research filed forward.

- Our proposal only considers the potential benefit of enabling the receiver's intelligence to alleviate the out-of-sequence data arrival problem in *MPTCP*. How to fully exploit the receiver's intelligence to intelligently handle the sender/receiver buffer overflow and mitigating the unnecessary retransmissions caused by dup-ACKs is an interesting topic worth further study.
- A GoP with the structure of IPBBPBBI is used in our simulation. In fact, in addition to the IPBBPBBI-structure, a GoP with less B-frames (e.g., IPPP or IBPBPPB) is often used in most of the video encoders. How the behaviors of our proposal with an arbitrary GoP structure (with less B-frames) is also worth further investigation.
- We argue that placing some *MPTCP*'s operations on the receiver, which is adjacent to the last-hop of wireless link, can achieve many and attractive benefits. However, adding some *MPTCP* operations on the client-side or receiver side may not be an option due to the diversity of receiver devices. The emerging Software Defined Networking (SDN) technology may be a good candidate to counteract the diversity of receiver devices and is worth of attention.
- It should be noted that the *MPTCP* technology is oriented to loss-sensitive more than time-sensitive applications. Applying a partial order service to the connection-oriented and fully reliable transport technology such as *MPTCP*, is still in controversy. The authors encourage more researchers to pay attention to and discuss this controversial issue.

VII. CONCLUSION AND FUTURE WORK

Motivated by the fact that the multimedia streaming services tend to be the world's most attractive service, furthermore, today's mobile devices are commonly embedded with more than one interface, this paper proposes a novel receiver-assisted partial reliable multipath transport solution

called as *recPR-MPTCP* for real-time multimedia streaming services. The *recPR-MPTCP* is an extension of *MPTCP* which runs the partial reliability operations at receiver side in order to (i) provide a partially reliable transmission scheme to *MPTCP*, and (ii) abandon an expired message at receiver rather than waiting for the decision from the sender. In addition, *recPR-MPTCP* provides a new receiver-based bandwidth aggregation method which uses the first-hand one-way forward delay as the performance metric to select a set of suitable path for multipathing. The experimental results show how the proposed *recPR-MPTCP* offers better quality of service for the transmission of both FTP traffic and multimedia traffic than the baseline *MPTCP* scheme.

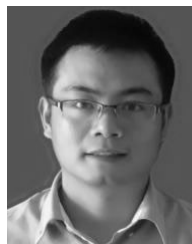
Although exploiting the receiver's intelligence can achieve some nice performance advantages in wireless multipath transmission, moving some operations of *MPTCP* from the sender-side to the receiver or client side will undoubtedly increase the energy consumption of a wireless mobile device [55], [56]. How to possibly reduce the energy consumption when running our proposal on a power-constrained mobile device is the important task of our future work. Moreover, our future work will consider applying the "smart collaboration" concept [57], [58] to optimize the performance of our *recPR-MPTCP* solution. To this end, all the clocks should be perfectly synchronized, and this could be a really hard problem [59]. We will consider synchronizing clock by using Network Time Protocol (NTP) [60], or synchronizing by using GPS satellites in our Linux-based hardware test-bed. Then the mandatory synchronization accuracy will be investigated, a benchmark video sequence with higher frame rate will be included, and the comparison with the receiver-centric multimedia-oriented *MPTCP* variants will be presented in the performance evaluation.

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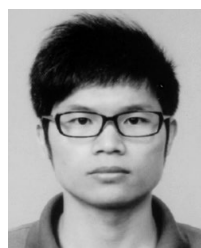


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