A MPTCP-based RTT-aware Packet Delivery Prioritisation Algorithm in AR/VR Scenarios

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***Abstract*—This work proposes, describes and performs performance analysis of a Round-Trip Time (RTT)-aware packet delivery prioritisation algorithm (RDPA) for networked Augmented Reality/Virtual Reality (AR/VR) content distribution. In this approach, the proposed algorithm uses the built-in multipath delivery feature of MPTCP. RDPA tracks, identifies and redirects the priority packets through the subflow that presents best opportunities to deliver the content with the lowest latency in the next transmission interval. This subflow selection is based on a linear regression that analyses each subflow behaviour and identifies the best subflow in terms of latency. The assessment of this algorithm is performed in a Network Simulator 3-based simulation environment and indicates performance improvements varying from 8% to 36% (peak performance) when compared with the MPTCP default operation.**

***Index Terms*—MPTCP, RTT-aware algorithm, prioritised content delivery, AR/VR**

I. Introduction

The Augmented Reality (AR)/Virtual Reality (VR) technologies have been gaining increasing market significance in recent years. Their applicability expands in many areas, such as the use of AR/VR as an enhancement tool for education and support for students with special needs [1] and in medical diagnostics for non-invasive data collection and rendering in real time [2] or therapies such as cognitive rehabilitation [3].

Additionally, its economic relevance projects a potential international market of US$80 to $150 billion by 2020 [4], [5]. This relevance can be perceived by the mobilisation of the academic sector (e.g. Harvard AR/VR [6], VR @ MIT [7]), big tech players (such as Microsoft's HoloLens [8] and Facebook's Oculus-Rift [9]) and many other companies (e.g. in the Khronos international consortium alone there are more than 100 companies working on royalty-free and open standard APIs for AR/VR [10]).

Nevertheless, as the use of AR/VR increases, new concerns about network infrastructure raise. The current networks offer significant enhancements for rich content delivery, however big challenges to supporting AR/VR content distribution lay ahead. For example, some works [11], [12] mention a demand of up to 5.2 Gbps bandwidth per user and an end-to-end round trip time (RTT) of one millisecond, whilst 5G technologies target 300 Mbps bandwidth and RTT of 10 ms [13].

This paper proposes an alternative solution to address existing network infrastructure resource limitations. This alternative explores the delivery prioritisation of AR/VR specific components, such as Inertial Measurement Unit (IMU), Global Positioning System (GPS) or infrared tracking data [14].

This paper introduces a new RTT-aware packet delivery prioritisation algorithm (RDPA), which is deployed on the top of the Multipath Transmission Control Protocol (MPTCP) and uses MPTCP's subflows for prioritised and regular data delivery. RDPA monitors, identifies and performs packet delivery using the MPTCP subflow that can offer the best RTT (latency) for the prioritised content. RDPA enhances a previous work on a Quality of Service "On-the-fly" algorithm (QoSF) which also focuses on RTT-based data delivery [15]. RDPA introduces two new aspects to improve the RTT-aware performance analysis: validation of each subflow TCP window availability (*subflow eligibility*) and monitoring of each subflow average RTT value (*subflow performance*). RDPA is modelled and assessed using the Network Simulator v3 (NS-3) and a NS-3 open source MPTCP implementation [16], based on the Internet Engineering Task Force (IETF) standard RFC 6824 [17].

This article is organised as follows. Section II presents the related works. The proposed RDPA algorithm is described in Section III. The simulation-based environment and test scenario are presented and testing results are analysed in Section IV. The conclusion and plans for future works are described in Section V.

II. Related works

The proposed RTT-aware algorithm works on top of MPTCP which operates at the transport layer of the Open Systems Interconnection (OSI) network stack and is implemented on the top of an open-source MPTCP implementation NS-3 [16] - which follows the IETF RFC 6824 [17].

The main objective of this algorithm is to analyse the subflows' RTT behaviour and find the best subflow to deliver prioritised packets in an AR/VR content delivery scenario.

In this context, the related work about technologies applied in or pertinent to this work, their applications, performance considerations and eventual limitations are presented. The related research works are divided in two subsections: *MPTCP Overview and Usage* and *AR/VR Content Delivery*.

A. MPTCP Overview and Usage

MPTCP enables data transport over multiple paths concurrently and transparently [17] at the transport layer of the

OSI stack. MPTCP manages multiple communication paths similar to TCP sessions created in parallel, which are known as MPTCP "subflows". MPTCP operates these subflows so they behave like regular TCP connections [17]. By not altering how the communication between OSI layers is established, MPTCP can be seamless integrated in the OSI model. The MPTCP stack is illustrated in Fig. 1, as discussed in [17], [16], [18].

The work of Hunger et al. [19] develops a redundant MPTCP (rMPTCP) packet scheduler targeting optimal fail-over time and smoother latency variations, reducing retransmission and improving network performance. Its approach to latency variation reduction differs from the algorithm proposed in this paper and is still subject to further tests to measure the trade-off between extra traffic load and latency variance reduction. Nevertheless, it corroborates that latency variations can be addressed in an MPTCP scenario.

Employing MPTCP for multimedia content transport, Corbillon et al. [20] have proposed an approach that explores the interaction between the transport and application layers (a cross-layer scheduler) and shows an improved performance for video streaming.

Once again, its approach differs from the work presented in this paper, but shows that managing the RTT can be a valuable tool for loss probability of each path. Despite that, the Corbillon et al. [20] cross-layer approach can face the problem of lack of support in middleboxes - which is guaranteed in the MPTCP IETF RFC 6824 [17], but comes backs when its use involves cross-layer communication.

The authors of [21] analysed how actual MPTCP implementations perform and showed that the MPTCP congestion control used does not obviate the need for a better subflow management process. Additionally, their reasoning indicates that an RTT-aware scheduling can offer limited benefits once window control mechanisms already account for RTT because it can somehow amplify path heterogeneity and, by doing so, it would have limited benefits. Nevertheless, the research presented in this paper demonstrates that an RTT-aware algorithm can improve the performance even though it sits on the top of the typical MPTCP scheduler.

B. AR/VR Content Delivery

Extensive efforts have been put proposing solutions able to deliver different types of rich media content at high quality across various networks. Some employed adaptation based approaches [22], others prioritisation [23]; some considered energy or cost as constraints [24], [25], others user perceived quality [26]; finally some focused at application layer [27], other at transport or lower layers [28] or even employed cross-layer approaches. These solutions are generic and could be applied for AR/VR, but they were not designed specifically for such content.

The work of [29] has an extensive exploration of many aspects related to 5G challenges to support AR/VR experience in a software-defined networking (SDN) architecture.

The authors have focused on the development of a multipath cooperative route (MCR) scheme, combining multipath alternatives and an intelligent distribution of content to multiple edge data centres (EDC). The distribution described in [29] takes into consideration the volume of data and the user proximity to a specific EDC. It also includes the use of a buffering policy to reduce the effect of delay jitter potentially present in a multipath and/or MCR implementation. The MCR proposed in [29] addresses the latency problem because AR/VR enables user interaction and, thus, the technology becomes latency sensitive. Notwithstanding the proposed innovative approach, it does not address the network delay directly, working on compensation or architectural alternatives.

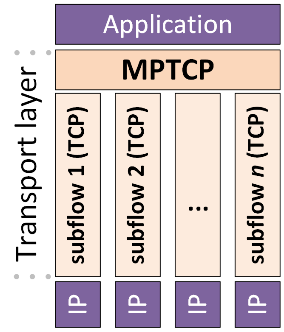

Figure 1: MPTCP stack

In another work that seeks alternatives for the latency problem, the authors of [30] explore the concepts of Mobile Edge Computing (MEC), the use of mmWave communication and the implementation of a proactive caching to deal with the AR/VR applications' stringent latency and reliability constraints. Using a VR gaming scenario, where players use wireless mmWave head-mounted VR displays (mmHMD), they simulate an environment where players could move freely in a mapped area. In that scenario, the authors develop a study that demonstrates the performance gains and the underlying trade-offs inherent to wireless VR networks. Although the scenario may not represent a real mobile condition, it still proves how the latency can be a serious limitation to the Quality of Service (QoS) or Quality of Experience (QoE).

While a significant number of academic works address the latency/delay, generically speaking, in an architectural way (multipath or otherwise), the stance adopted by this paper explores it at the protocol level, i.e. it explores potential opportunities brought by the MPTCP at the transport layer level.

Finally, despite the fragility of formal standardisation for AR/VR technologies as a whole [31], [32], it is noticeable that recent efforts have been made to address this problem. Both IEEE [33], [34], [35] and VESA [36] announced, in May 2017, the formation of special groups and projects focused on the development of standards for AR/VR.

III. MPTCP-based RTT-aware Packet Delivery Prioritisation

This paper examines how MPTCP multipath characteristics can be explored to improve the delivery performance of prioritised packets during AR/VR content delivery.

AR/VR solutions become increasingly complex and resource demanding (bandwidth, latency, etc). In the last few years, solutions varying from a Google®-like Cardboard to Facebook® Oculus series, in Fig. 2, are examples of how fast the technology is evolving and its adoption is under private and academic scrutiny [37].

Not only the video bandwidth has more stringent demands, considering the increasing quality of the videos (UHD and beyond, 360 videos, etc), but other types or streams of information also have high demands - especially when user interaction is essential to the overall experience [38], [39]. Therefore prioritisation is so important for AR/VR applications and this aspect must be addressed accordingly.

Despite the fact that the type and size of the prioritised data (e.g. IMU, GPS, joystick or motion trackers [14]), as shown in Fig. 2, are considerably smaller, it demands a higher priority in terms of timely delivery when compared with other typical AR/VR components (e.g. video data components).

In the context of this AR/VR scenario, this paper presents an RTT-aware algorithm that employs the MPTCP's subflow features to enable content delivery prioritisation.

A. The RTT-aware Packet Delivery Prioritisation Algorithm (RDPA)

RDPA makes use of the fact that there are different MPTCP subflow characteristics and selects the most appropriate subflow for the prioritised data delivery in order to achieve better delivery performance in terms of latency. This is performed without altering MPTCP's basic TCP-like operation [17]. In this process RDPA will make use of MPTCP's default load balancing algorithm.

RDPA improves a previous work in which a Quality of Service algorithm (QoSF) was introduced with a similar RTT-aware behaviour [15]. RDPA introduces two new aspects to enhance the RTT-aware performance analysis.

First, for each MPTCP subflow RDPA validates the TCP window available for the transmission before evaluating the RTT historical behaviour. This avoids wasting time with further calculations when the inappropriate TCP window size would make impossible any packet transmission and as result the data would be returned to the default MPTCP load balancing algorithm.

Second, RDPA takes into consideration the RTT mean value of the given subflow historical behaviour. This value helps avoid subflows with small RTT slopes and yet high RTT trend. It is represented by θ in Equation (1).

RDPA has three distinct stages: monitoring, subflow selection and content delivery.

RDPA monitors the RTT behaviour distributed in the subflow pool (historical data) and prioritised packet detection. It also monitors each subflow transmission in the subflow pool to track the RTT behaviour for future calculations.

For subflow selection, when the RDPA detects a prioritised packet, its algorithm calculates and predicts which subflow has the highest chances of providing the best delivery performance based on their lowest latency.

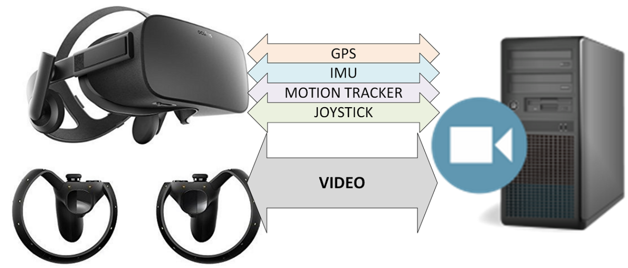

Figure 2: MPTCP stack

After the "best fit" subflow is defined, RDPA alters the MPTCP default load balance and redirects the prioritised packet using the selected subflow.

The proposed RDPA algorithm is illustrated in Fig. 3 (as an extension of the MPTCP stack presented in Fig. 1) and the Algorithm 1 describes in detail how it chooses the subflow based on the subflows' historic RTT behaviour.

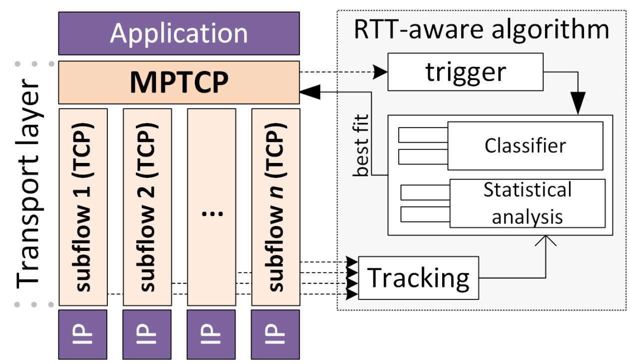

Figure 3: RTT-aware algorithm

The subflow selection mentioned above is based on a linear regression calculated for each subflow. For that, the smaller the subflow's linear regression slope is, the better are the chances it will deliver the lowest latency in the next operation. The linear regression slope equations is presented in Equation (1).

$$S = \frac{n\left(\sum_{i=1}^{n} x_i y_i\right) - \left(\sum_{i=1}^{n} x_i\right)\left(\sum_{i=1}^{n} y_i\right)}{n\left(\sum_{i=1}^{n} x_i^2\right) - \left(\sum_{i=1}^{n} x_i\right)^2} * \frac{1}{\theta}$$

$$\theta = \frac{1}{n}\left(\sum_{i=1}^{n} y_i\right) \tag{1}$$

$$S(i) \in \mathbb{R}, \forall i = 1, ..., \mathbb{R}$$

Where S is the linear regression's slope indicating the trend for the next value of RTT, n is the number of samples used, x_i represents the i_{th} value of x (time value), y_i represents the i_{th} value of y (RTT value) and θ is the RTT mean value of the given subflow. The domain of $S(i)$ belongs to the set of real numbers $\mathbb{R}$ and all values of i are in the domain of the natural numbers, $\mathbb{N}$.

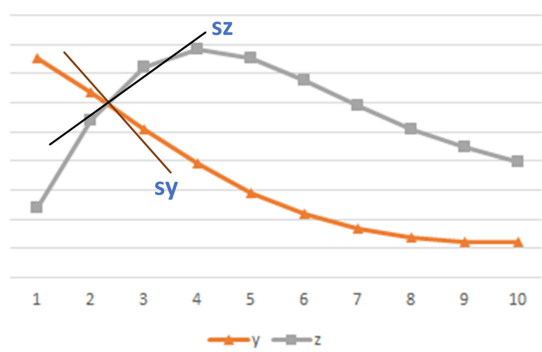

Figure 4: RTTs' linear regression (hypothetical)

A hypothetical example is shown in Fig. 4. Two subflows' RTT historical data (y and z) are analysed when a prioritised packet is detected. At the moment of the prioritised packet detection, both subflows have the same RTT value.

The linear regression slope for each subflow is calculated (sy and sz) and the smallest slope (sy, in this case) indicates the subflow with best chances to offer the lowest latency.

However, an extra condition must be observed. The TCP transmission window must be wide enough to transmit the urgent packet, as generically defined in Equation (2).

$$SW(s) = (A(s) - P(s)) >= 0$$
$$P(s) <= A(w) <= W(q) \tag{2}$$

where SW(s) validates if there is enough TCP transmission window for a subflow, $P(s)$ is the packet size to be transmitted, $A(s)$ is the available TCP transmission window for specific subflow and $W(q)$ is the total TCP transmission window size.

IV. Simulation Testbed

The proposed RDPA algorithm is evaluated in a simulation environment built based on the Kheirkhah et al. [16] NS-3 open source MPTCP implementation of the IETF RFC 6824 [17]. The simulated topology is shown in Fig. 5.

For the assessment, a point-to-point network model is employed, with all nodes in the topology connected using links with 1 Mbps data rate and 2 ms delay. Despite being a reduced-complexity simulation testbed, as can be seen in Fig. 5, this topology still accommodates realistic results for the analysis of the proposed RDPA performance.

In this model, a MpTcpBulkSender application is set on n_0 and a MpTcpPacketSink application is set on n_4, using a single-homed configuration - where one device is available on each node and 8 subflows are established using different ports. These applications are extensions of the standard applications found on NS-3 and are designed to send and receive simulated AR/VR data as fast as possible using MPTCP.

In this scenario, three streams of data were considered, associated with IMU, GPS and video components, respectively. According to the work of [14], the ratio between IMU and GPS data which are carried by prioritised packets and video data packets is 1 to 500.

Algorithm 1: RTT-aware algorithm.

```
Result: Dynamically tracks, identifies and sends the
        packet using the best subflow available.
Input:  PKT ← current Packet being transmitted
        SID ← current subflow in used
        HiST ← subflows' RTT history info
if (isPriority(PKT)) then
    bestValue = ∞
    mode = 0; //A queue stores the history of a subflow
    foreach Queue in HiST do
        //Check available transmission window
        if (Queue.window - PKT.size ≥ 0) then
            S, sX, sX2, sXY, b, sY = 0
            n = Queue.size()
            //Calculate the linear regression factors
            foreach obj in Queue do
                sX += obj.time
                sX2 += pow(obj.time, 2)
                sY += obj.RTT
                sXY += (obj.time * obj.RTT)
            end
            //Calculate the linear regression slope (S)
            numerator = ((n * sXY) - (sX * sY))
            denominator = ((n * sX2) - pow(sX, 2))
            S = (numerator / denominator) * (sY / n)
            //Check for the smallest slope
            if (S < bestValue) then
                SID = Queue.subflowId
                bestValue = S
            end
        end
    end
end
send(PKT, SID) ← Sends packet using chosen subflow
HiST.Queue(SID).add(PKT.time, PKT.RTT) ← Tracks
```

The MpTcpBulkSender is configured to send simulated AR/VR content data as fast as possible over the network using MPTCP and RDPA on top of MPTCP, respectively. Tests last a period of 1200 seconds and no particular background traffic which could interfere with the transmission is considered. The first 200 seconds of this tests are considered a transient state and are not detailed in this paper. Table I has a summary of the settings used in this testbed.

A. RDPA Performance Assessment

The results in Table II are a sample of the RTT measurements for the prioritised packets on MPTCP default operation and on the proposed RDPA. These results can also be seen in Fig. 6 where a steady period of the test is analysed. When comparing the use of the proposed RDPA and MPTCP, in terms of RTT (latency) performance, RDPA outperforms the MPTCP default operation by more than 8% (and a 36% peak).

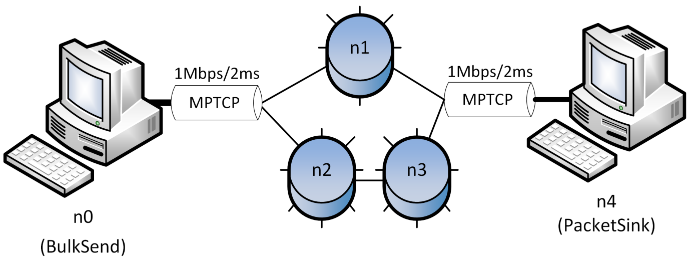

Figure 5: Testbed topology

Table I
SIMULATION SETUP SUMMARY

Parameter	Value
Environment	NS-3 open source MPTCP [16]
Simulation length	1200 seconds
Number of nodes	4 Nodes
Data Rate	1Mbps
Delay	2ms
Number of subflows	8 subflows
Prioritised/non-prioritised ratio	1/500
Sender app	MpTcpBulkSender [16]
Receiver app	MpTcpPacketSink [16]

Additionally, RDPA has also a slightly better throughput performance when compared with MPTCP. In this testbed, the results indicate a 3.6% increase on average.

The RTT variations between the MPTCP (non-prioritised) and RDPA (prioritised) RTT performance, illustrated in Fig. 6, can be summarized in segmented periods of time, as shown in Fig. 7. This offers a high-level visualisation of how RDPA outperforms the MPTCP operation. It also demonstrates that the gains are not only a transient phenomena, but a steady characteristic in MPTCP and RDPA operations.

Finally, considering eventual side effects caused by the implementation of this RTT-aware algorithm, other aspects of TCP protocol are monitored.

As shown in Table III, there is no significant degradation in retransmission, duplicate ACK, lost segment or fast retransmission when comparing MPTCP and RDPA.

The number of retransmissions has a marginal 0.1% increment and, according to [40], all those values are yet considered acceptable for most application types. The stability in these parameter levels, especially for retransmissions and lost segments, means that the RDPA implementation has no side effect that would lead to any delays and increase the latency - what would deteriorate the QoE and, in some cases, may be responsible for a "motion sickness" effect.

V. CONCLUSIONS

This paper proposes a new RTT-aware algorithm for prioritised AR/VR data delivery (RDPA), built on top of MPTCP. RDPA evaluates all subflows' performance and selects the one with the lowest latency for priority data transport. RDPA performance is compared with the MPTCP default (non-prioritised) operation.

Table II
RDPA PERFORMANCE - SINGLE-HOMED CONFIGURATION

MPTCP			RDPA		
time (s)	RTT (ms)	thru (B)	time (s)	RTT (ms)	thru (B)
204.9	1005	1393	199.1	1017	1377
210.7	1052	1331	205.0	1029	1361
216.5	1087	1288	210.8	1040	1346
222.4	1099	1274	216.6	1064	1316
239.9	1146	1222	234.1	1099	1274
245.7	1157	1210	240.0	1111	1260
251.5	1192	1174	245.8	1122	1248
...	...	...	...	...	...
1264.6	1146	1222	1270.9	994	1408
1270.4	1146	1222	1276.8	1017	1377
1276.2	1146	1222	1282.6	1052	1331
1282.1	1169	1198	1288.4	1064	1316
1287.9	1169	1198	1294.2	1087	1288
1293.8	1181	1185	1300.1	1111	1260
1299.6	1192	1174	1305.9	1134	1235
1305.4	1192	1174	1311.7	1181	1185
Average RTT					
1101ms ± 87ms			1015ms ± 99ms		
Average Throughput					
1281B ± 117B			1327B ± 136B		

Table III
PACKET LOSSES - SINGLE-HOMED

Type	MPTCP (%)	RDPA (%)
retransmission	1,07	1,17
duplicate ACK	2,54	2,59
lost segment	0,47	0,46
fast retransmission	0,02	0,01

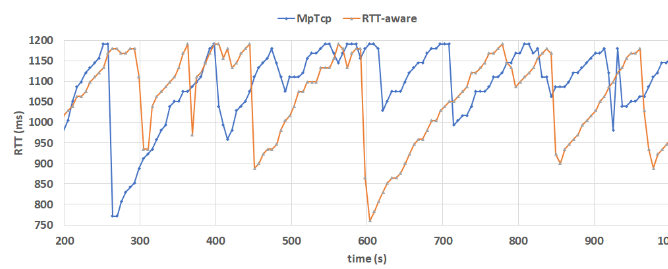

Figure 6: RTT comparative results for RDPA and MPTCP

The NS-3 simulated environment tests indicate RTT performance gains of more than 8% (and a 36% peak). Additionally, the results indicates a throughput performance improvement of about 3.6% on average.

Future investigations should adapt and assess the proposed RDPA in other aspects of prioritised AR/VR delivery, including AR/VR video and/or audio components delivery, QoE assessment and other techniques related to AR/VR delivery.

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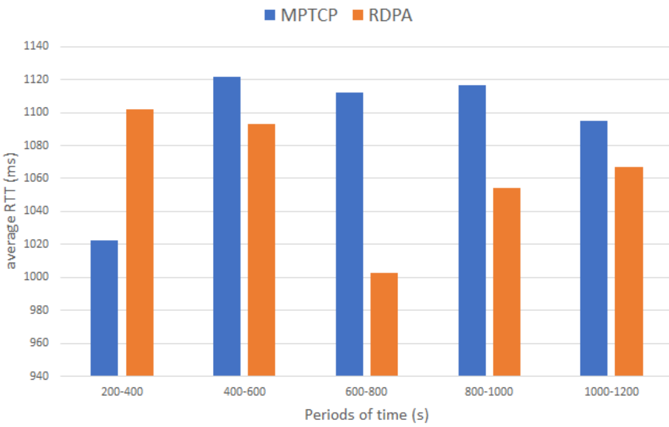

Figure 7: Steady analysis for RDPA and MPTCP (average)

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