TCP Wave estimation of the optimal operating point using ACK trains

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Abstract—TCP Wave changes the typical TCP transmission paradigm by replacing the ACK-clocked sliding window with self-scheduled bursts. In response to bursts, unmodified TCP receivers generate ACK trains, which carry useful information about the end-to-end link characteristics. TCP Wave inspects ACK-train flow to measure the following parameters: (i) ACK train spread (namely ACK train dispersion) and (ii) RTT variations. The former is expected to provide the overall “service capacity”, meant as the maximum capacity allowed over the end-to-end path, while the latter is considered as a congestion indicator. The joint use of such measurements allows TCP Wave to fine-tune the transmission rate to accurately match current network resource availability in the bottleneck link. This paper analysis TCP Wave ACK train-based measurements on a broad set of simulated links compliant to characteristics of today’s real networks. To this scope, a testbed with TCP Wave implementation on Linux OS is used to perform tests varying both bottleneck capacity and physical latency.

Keywords—TCP Wave; bandwidth estimation; Linux; bottleneck

I. DESCRIPTION AND LIMITATIONS OF TCP CONGESTION CONTROL

In traditional TCP [1], Congestion Window (*CWND*) value updates and scheduling of new segments to be transmitted on the network are triggered upon reception of ACKs, making TCP transmission in fact ACK-clocked. On the other hand, ACK reception timing is depending on the Round-Trip Time (RTT), computed at the sender, as the time interval between a packet sending and the reception of the corresponding Acknowledgement (ACK). Thus, the higher is the RTT, the slower is the triggering of TCP rate adaptations, which can be realized by different congestion control algorithms such as New Reno, Cubic, Vegas, Westwood, etc. Furthermore, TCP generally follows a reactive strategy for the congestion control: *wait for the trouble and then solve it*. Congestion control algorithms progressively increase *CWND* upon ACK receptions and then reduce it in case of loss detections (through duplicate ACK receptions or Retransmission Time Out expiration RTO). In both cases, operation timing is entirely relying on RTT. The resulting protocol was performing quite good in 80s-90s wired networks characterized by:

- Low RTT, allowing fast actions on *CWND*;
- Relatively small bandwidth (BW) at the bottleneck, keeping small the *CWND* value to achieve maximum rate (provided by the BW·RTT product).
- Small bottleneck buffers, limiting the RTT upper bound;

- Low occurrence of link transmission errors, so that all losses are most likely due to congestion (justifying the *CWND* reduction).
- Long transfers, providing time to achieve TCP steady-state.

Such a communication environment went through significant changes over the years, with the continuous increase in link bandwidth, the proliferation of wireless links and/or lossy links (with transmission errors) [2][3], the higher in-network buffering capability [4][5], and the deployment of hybrid networks with variable topologies and physical characteristics [6]. Besides, HTTP traffic dominance led to frequent short transfers [7], usually terminated before the achievement of the TCP steady-state. All the above factors contribute to break original TCP design assumptions leading up to significantly impaired performance. Consequentially, protocol enhancements and modifications became needed with the aim to allow transmissions at the maximum eligible rate (optimal bottleneck utilization) without significant increases on the end-to-end RTT (no congestion and flow fairness). Such a condition is referred as *optimal operating point* and cannot be achieved by traditional *CWND*-based and ACK-clocked TCP paradigm, because it leverages on the RTT only. In fact, the convergence to the *optimal operating point* is a problem with two-variables: the *Bottleneck Capacity* (BC) and the *Congestion Level* (CL). A single sender-side measurement, such as the RTT, is then not sufficient to assess the optimal transmission settings. For instance, RTT increases can be jointly or alternatively due to both a rate higher than exceeding and CL increases (more competitive traffic), and it is quite challenging to solve this ambiguity. An additional sender-side measurement is then necessary to solve this problem. At the same time, another limitation can derive from the ACK-clocked operations of standard TCP, which limit the reactivity to changes being proportional to the RTT.

TCP Wave is a recent alternative transport protocol that pursues by design the objective to work around the *optimal operating point*, by leveraging on two “not-correlated” sender-side measurements, which are allowed by a new transmission paradigm based on bursts (which are not ACK-clocked). In fact, transmission in bursts allows the generation of so-called *ACK-trains*, which carry, together with traditional RTT information, the ACK train dispersion over time, strictly correlated to the BC. TCP Wave modifies the congestion control mechanisms, without requiring neither ad-hoc methods at the lower layers (i.e., ad-hoc packet scheduler) nor receiver changes. Therefore,

it can constitute a promising new technology once demonstrated the validity and consistency of the performed measurements over a broad range of network configurations.

In this framework, the scope of this paper is to validate the TCP Wave measurements with respect to the expected (ideal) values, using a real protocol implementation on Linux kernel [8]. Specifically, Section II provides a general overview of TCP Wave aimed to highlight how ACK-based measurement impact on protocol functions; Section III details the method to take the target measurements and how they are used to compute a suitable $TxTime$ applied for burst transmission; Section IV presents result of a test campaign tailored to the verification of the measurement validity with some preliminary performance results, postponing a full characterization of the rate control performance in future works.

II. TCP WAVE: PROTOCOL DESCRIPTION

TCP Wave is a transport protocol designed as an alternative to traditional *CWND*-based TCP versions for current broadband networks. A full protocol specification is provided in [9], while performance assessments under various simulated network and application configurations are also studied in [10][11][12][13].

The main innovative protocol characteristics can be summarized as follows:

- Burst transmission paradigm;
- A transport layer self-managed clock at the TCP sender;
- Proactive rate control.

The joint effect of such characteristics leads to a reliable transport protocol radically different from traditional TCP design, while still guaranteeing the fundamental TCP properties: congestion control and avoidance, flow control, error control, inter-flow fairness and friendliness. Therefore, TCP Wave is not an additional TCP enhanced version, but it proposes a reviewed transmission paradigm compliant to the fundamental TCP principles. At the same time, the transport layer receiver does not need any modification, making possible a fast deployment of the new protocol in co-existence with current TCP implementations.

The fundamental principle of TCP Wave is the packet transmission in bursts adequately spaced each other. Thus, scheduled transmissions envisage bursts of BURST packets, with a default value BURST₀ of 10 packets (used as the initial value and as a target value during the connection). In practice, TCP Wave sender increases the CWND by BURST upon the expiration of an internal timer, which has an expiration time equal to a value called $TxTime$, so that the actual transmission rate is given by:

$$TxRate = \frac{BURST}{TxTime} \quad [Eq. 1]$$

Then, TCP Wave transmission rate is fully regulated by the current $TxTime$ value, while ACK reception does not trigger any direct action on the CWND. $TxTime$ determines the actual rate and its value is dynamically updated in accordance with both *BC* and *CL* measurements. In particular, updated $TxTime$ values are computed by an *ACK-based Capacity and Congestion Estimation* (ACCE) component that continuously monitors the received ACK flow with the aim to measure two parameters: the

RTT (indicator for *CL*) and the ACK train dispersion (indicator for *BC*).

Waiting for the reception of the first ACK train (corresponding to a full BURST transmission), $TxTime$ is set to a default value ($TxTime_0$), which determines the rate during the initial *blind phase*. Most likely, such a phase lasts 1 RTT only. Recommendations to define target BURST₀ foster the achievement of the most of shortest transfer (i.e., messaging-like exchanges) within the first burst, while $TxTime_0$ is set to guarantee a minimum transmission rate to short transfers needing more than one burst, but terminating before the reception of the first ACK train.

Another key change introduced by TCP Wave concerns the attempt to strictly follow the network status and set the transmission rate to prevent congestion. TCP Wave logic is to transmit “as much as possible” without inflating the RTT, then constantly working around the *optimal operating point*. This is in contrast with the traditional TCP “reactive” philosophy: “send even more packets until losses are detected, then repair from both losses and injected congestion”.

Upon the reception of the first ACK train, TCP Wave enters in the *Tracking Mode*, in which the $TxTime$ is updated by any further ACK train reception using the following equation:

$$TxTime_n = ACK_Train_Disp_n + 0.5 \cdot \Delta RTT_n \quad [Eq. 2]$$

where n indicates the ACK train sequence number. The ACK train dispersion factor (*ACK_Train_Disp*) indicates an assessment of the $TxTime$ to achieve *BC*, while $0.5 \cdot \Delta RTT_n$ is a correction factor that takes into account *CL*. In more details, ΔRTT_n is the difference between the current average RTT_n and the minimum measured RTT (RTT_{min}). The 0.5 multiplier factor is to equally split the charge for the congestion recovery between the target TCP Wave connection and the competing flows. In theory, a TCP Wave connection should run in *Tracking Mode* for its whole lifetime. A ΔRTT threshold, namely β , is defined to take into account situations where RTT increase is significant, despite the continuous rate adaptation following [Eq. 2]. A significant increase can occur in various circumstances: greedy competing traffic (on top of different transport protocols, i.e., UDP), network changes, an unusual high number of simultaneous flows creating a temporary congestion. When $\Delta RTT_n > \beta$, TCP Wave triggers *Adjustment mode* that performs a reset of both the transmission rate (reverting $TxTime$ to $TxTime_0$) and the RTT_{min} reference. The rationale is to temporarily transmit at a minimum rate and sense if something is changed in the network by looking at new ACK trains. In order to skip the information coming from the data in-flight when the algorithm enters *Adjustment Mode*, a “Stability Factor” value is calculated, following [Eq. 3].

$$StabilityFactor = \frac{RTT_n}{TxTime_n} \quad [Eq. 3]$$

In this way, a new $TxTime$ update, which is the event to go back in *Tracking Mode*, is computed after skipping an upcoming number of ACK trains equal to *Stability Factor*. As a consequence, the duration of *Adjustment Mode* is roughly RTT_n .

III. ACCE MEASUREMENTS

TCP Wave operations leverage the assumption that measurements provided by the *ACCE* component are reliable and accurately reflect actual *BC* and *CL*. This requires that *ACK Train Dispersion* reflects the ideal BURST spacing due to the bottleneck [14] for the achievement of a transmission rate equal to *BC*, and the RTT increases reflect the actual congestion status filtering possible spare traffic spikes, the effects of possible retransmissions, cumulative ACKs, ACK/packet aggregation, and so on. Therefore, procedures and algorithms applied in taking the measurements are of paramount importance.

A. ACK flow handling

ACCE monitors the whole ACK flow to compute target statistics: the RTT of the first packet of each burst, namely RTT_{pilot} , and the time between last ($last_ack_time$) and first ACK ($first_ack_time$) of a train. As a reference, it has access to the burst history queue that records the size of in-flight bursts (transmitted and waiting to receive the corresponding ACK train). As soon as a burst is fully acknowledged, a measurement round is completed and the corresponding record is deleted from the burst history queue. Therefore, the first-placed record into the queue provides to *ACCE* the boundaries of the current ACK train.

Basically, a new ACK can trigger a set of operations associated to three logic states managed by the *ACCE* algorithm:

- **Round_begin** – it is triggered when ACK is believed as the first ACK of a train. This ACK is used to compute the RTT value associated to the whole burst (RTT_{pilot}) and to set the initial time reference for the current ACK train ($first_ack_time$).
- **Round_middle** – it is triggered for each ACK received following the first ACK in the train. It updates both the number of ACKs within the train and the time reference of the last ACK in the train ($last_ack_time$).
- **Round_end** – *ACCE* enters in this state when received ACK completes the currently monitored train. Basically, the ACK updates $last_ack_time$ and triggers final calculation of both ACK train dispersion and the RTT values.

Upon an ACK reception, both the number of acknowledged bytes (*ACKed bytes*) and the RTT (current RTT) are preliminarily examined before entering a logic state. Because a consistent ACK train structure requires more than one ACK per burst, *ACCE* filters any “big” cumulative ACK that would not allow train-based measurements. When *ACKed bytes* are larger than the size of the current burst, ACK train structure is considered broken and any timing parameters previously registered for the current train not consistent. Accordingly, any fully acknowledged burst by such a “big” ACK is cleared from the burst queue, while all the timing information is reset. This reset operation allows the TCP Wave transmission parameters to be updated from the latest correctly computed measurements instead of relying on inconsistent samples. Whereas, if there are no valid measurements previously achieved, the transmission parameters are simply not updated and $TxTime_0$ remains effective. Still, the “big” cumulative ACK may also trigger the **Round_begin** if it partially covers a burst in the queue (newest

burst which is partially acknowledged by the ACK), making the ACK as the first one of the new train.

Without TCP losses, ACKs usually cover a number of bytes lower than burst size (TCP receiver triggers one ACK for every two full segments by default [15]), and ACK train structure is kept consistent. Thus, *ACCE* proceeds by checking if the received ACK is the first one in a train. This is done by checking the $first_ack_time$ value, which is set to the current time by the **Round_begin**, and then reset to 0 when the **Round_end**.

Whenever an ACK is not the first of a train, a further check on its timing consistency is performed through the associated RTT value (*current RTT*). Since a burst transmission envisages the in-order injection of multiple segments at once (same sending time reference), the RTT within an unaltered ACK train must be monotonically growing. On the contrary, a decreased RTT indicates a misalignment between the ACK train and its corresponding burst. Thus, in this case, *ACCE* enforces a **Round_begin** because it considers the ACK as the first of the train. This occurrence is mostly due to packet transmissions by the underlying operating system out of the TCP Wave control, which create unexpected shifting in the ACK train structure.

In any other case, when ACK is neither marked as first of the train nor presents a decreased RTT, it is assumed to be in the middle of the current train, and therefore **Round_middle** is triggered. For each **Round_middle** ACK, *ACCE* tracks the sum of *ACKed bytes* by the train. If the *ACKed bytes* is greater than or equal to corresponding burst size *BURST*, the ACK is assumed to terminate the train, and the measurement for the current burst is concluded triggering a **Round_end**. However, within this last state, two sub-cases are possible due to cumulative nature of ACKs: (1) the ACK acknowledges the exact number of segments in the burst, or (2) the ACK also acknowledges some segments of the subsequent burst.

In the first case, information for the number of ACKs in the train, $first_ack_time$, and $last_ack_time$ are used to compute the ACK train dispersion and the RTT measurements for the current train are concluded. Consequently, all the internal *ACCE* variables are reset and the upcoming ACK will be accounted as the first of a new train. In the second case, the last ACK terminates the current train but also includes one or more segments of the subsequent burst. This means that the ACK carries timing information inflated by transmission of two consecutive bursts. Thus, *ACCE* uses this last ACK to terminates the round but utilizing the $last_ack_time$ set by the prior ACK in the previous **Round_middle**. Furthermore, a **Round_begin** must be triggered immediately taking into account the last ACK as the first of a new round.

B. Bottleneck capacity and congestion assessment

At the end of each round, *ACCE* aims to provide as outputs possible new samples for ACK train dispersion and RTT, feeding TCP Wave algorithms. A new ACK train dispersion sample is computed only if the burst size is equal to the reference burst size ($BURST_0$), because smaller trains could cause two main drawbacks for the measurements:

1. A normalization is required for the final ACK train dispersion sample used to update $TxTime$, which is always tailored for $BURST_0$ transmissions;

2. The smaller is the burst and the lower is the number of ACKs within a train, making the measurements mostly sensitive to timing glitches.

Then, when the size of the current burst is smaller than the $BURST_0$, ACCE returns the value of the latest valid measurement. However, transmission of burst smaller than $BURST_0$ can be due to sender buffer limitations, or application-limited data at the moment the burst is generated. Similarly, when the number of received ACK in a train is lower than 3, the ACK train dispersion is not updated.

On the contrary, when the current burst size is equal to $BURST_0$, a “fresh” ACK train dispersion is computed as follows:

$$ACK_Train_Disp_n = \frac{(last_ack_time - first_ack_time) \cdot \frac{number\ acks\ per\ train}{number\ acks\ per\ train - 1}}{1} \quad [Eq. 4]$$

The last term in [Eq. 4] adds a time contribution to be accounted after the last ACK, equal to the average spacing among the consecutive ACK pairs within the train. In fact, $last_ack_time$ does not include the time needed to process the last segment of the burst. Definitively, the calculated value for the ACK train dispersion is used to update the $TxTime$ value by feeding [Eq. 2].

The RTT measurement function takes as input argument only the RTT_{pilot} value, computed in the **Round_begin**. Such an RTT value is representative for the whole burst and it is used to assess congestion status. First, TCP Wave sender updates the minimum RTT experienced overall the current *Tracking mode*, namely RTT_{min} , as the lowest RTT_{pilot} value. Additionally, to avoid the effect of spare RTT spikes that may occur due to network dynamics not strictly correlated to the congestion status, a smoothed RTT, namely RTT_{avg} , is computed by using an Exponentially Weighted Moving Average (EDWA) filter:

$$RTT_{avg} = \alpha \times RTT_{avg} + (1 - \alpha) \times RTT_{pilot} \quad [Eq. 5]$$

where α is the filter parameter dynamically computed as:

$$\alpha = (RTT_{pilot} - RTT_{min}) / RTT_{pilot} \quad [Eq. 6]$$

Thus, the larger is the difference between the new RTT_{pilot} and the RTT_{min} , the stronger is the smoothing effect performed by the α filter.

The final output of the RTT measurement function is the ΔRTT value:

$$\Delta RTT = RTT_{avg} - RTT_{min} \quad [Eq. 7]$$

Such a value feeds [Eq. 2] and then jointly to the computed ack_train_disp , determines the updated $TxTime$. In addition, ΔRTT is compared to the β threshold, in order to determine the next TCP Wave mode, either *Tracking* or *Adjustment*.

IV. ACCE MEASUREMENT VALIDATION

As a target objective, ACCE is required to continuously compute updated estimations of BC in terms of ACK train dispersion and CL in terms of ΔRTT . The higher is the percentage of valid samples, the better is the supported reactivity to network and load changes. Thus, the percentage of

Round_end triggering represents a clear indication of the validity of the proposed scheme for handling ACK flow. In addition, the evaluation of the achieved ACK train dispersion samples with respect to the ideal value is of paramount importance for the measurement reliability. Inconsistent values may lead to wrong assumptions on the underlying network with a harmful impact on the transmission rate. By design, TCP Wave rate control uses ΔRTT to compensate a possible underestimation of the ACK train dispersion, making somehow the protocol robust to small measurements errors.

As an ideal benchmark, optimum performance for a single TCP Wave flow needs to achieve from ACCE ACK train dispersion such that:

$$BC = BURST / (ack_train_disp) \quad [Eq. 8]$$

and $\Delta RTT = 0$.

For this purpose, we built a basic testbed setup, shown in Figure 1, that consists of three Linux machines acting as sender, receiver and bottleneck link emulator, respectively. Each node pair is connected through 1 Gbit/s network ethernet interface. In the sender machine, we run a modified kernel that contains the latest implementation of TCP Wave [8], while receiver runs a default Linux network configuration. The intermediate node is configured as a Linux based router (ZeroShell) [16] in order to provide a traffic shaping and QoS management able to apply bandwidth and delay constraints.

Fig. 1: Testbed Setup

For the sake of an exhaustive test campaign, the bandwidth value is varied in the range [2 Mbit/s – 30 Mbit/s], with a 2 Mbit/s granularity. Such a range covers most of the current consumer IP connectivity services over public networks. Besides, the physical delay is jointly varied to reproduce four different communication scenarios in terms of end-to-end physical RTT: no extra delay ($RTT \approx 0.3$ ms), $RTT = 100$ ms, $RTT = 200$ ms, $RTT = 500$ ms. Note that ACK train dispersion should be independent of latency, so that it is worth to verify such a property over a large set of RTT configurations in real environments. Tests consist on running an Iperf [17] transfer of 500 Mbytes over a TCP Wave connection in order to gather a high number of ACK train dispersion samples (≈ 36 K), allowing to achieve a statistically significant *Probability Density Function* (PDF). Figure 2 shows the PDF of three selected bandwidth configurations from our test campaign. Figure 2 (a) shows PDF of the ACK train dispersion in case of BC equal to 2 Mbit/s in comparison with the optimum value (60.6 ms) to exactly match BC . First, results demonstrate how measurements are resilient to different possible RTT, because of the almost precise overlapping among the various curves. Simulation output reports that the overall percentage of valid samples (triggering a successful Round-end) varies from 80.6% with $RTT = 500$ ms to 99.9% with $RTT = 0.3$ ms.

Fig. 2: PDF of ACK train dispersion measurements

Therefore, ACCE is able to maintain a continuous synchronization with the ACK train structure efficiently feeding TCP Wave rate control component with ever updated values. Valid measurements fall in more than 80% very close, while slightly higher, to the optimum value. Definitely, the ACCE outputs strictly reflect BC and then represent a suitable input for TCP Wave rate control.

Figure 2 (b) concerns the test with BC = 10 Mbit/s. The number of valid samples is similar to that achieved in the previous case, then confirming a very good synchronization with the ACK train structure. Although the most of the measurement values are contained in a small-time interval around the ideal one (12.11 ms), PDF shape results a bit more spread over such an interval, as remarked by the PDF peak at about 0.35. This means that ACK train dispersion presents a larger standard deviation from the average value. Finally, PDF with BC = 30 Mbit/s, in Figure 2 (c), presents similar characteristics and leads to similar considerations as those discussed for BC = 10 Mbit/s. It is worth to remark that such a high bandwidth implies measurements of much smaller time intervals which are more sensitive to little variations from the ideal value (4.04 ms).

A. Preliminary TCP Wave performance analysis

Figure 3 summarizes performance achieved by TCP Wave in terms of average bottleneck utilization, during the same simulation runs described before.

Fig. 3: Measured goodput of TCP Wave over different BDP links

TCP sets transmission rate by updating $TxTime$ as defined by [Eq. 2]. Full utilization ($=1$) indicates a perfect matching between transmission rate and BC. Results in Figure 3 show a little utilization improvement as the latency decreases. This trend can be explained by the analysis of ACK train dispersion PDF curves that present some small tails at values lower than the average one in case of high RTT. This leads to slightly over transmit and to a proportional increase of RTT, which in turn equalizes the next $TxTime$ update. Such a control loop makes small oscillations of the transmission rate over time (which in fact give the name to the protocol), that result in the marginal utilization decrease. Utilization slightly decreases also for the highest bottleneck capacities. This is due to the handling of smaller ACK train dispersion measurements, for which some little deviations from the ideal value causes a higher rate reduction.

The achieved transmission rate depends on the measured ACK train dispersion as well as the computed ΔRTT which, for a complete analysis, whose values are reported in Figure 4 for the different configurations. The gathered low ΔRTT values indicate a transmission rate quite similar to the delivery rate, meant as the actual rate at the receiver, and no major self-induced congestion to the bottleneck is generated.

Fig. 4: ΔRTT measurements over different BDP links

As a summary, overall performance is optimal with a bottleneck utilization always above 92% and marginal values of ΔRTT (< 2 ms), which indicate how TCP Wave is able to work

in the proximity of the *optimal operating point* over a large range of network configurations

V. CONCLUSIONS

This paper investigates the consistency of ACK train-based measurements performed by TCP Wave to update its transmission rate. Tests leveraged a Linux TCP Wave sender implementation, which allowed to interwork with a real TCP receiver, then making ACK generation realistic. Results are very promising with filtered ACK train dispersion samples strictly reflecting the bottleneck capacity and then efficiently exploitable to implement the TCP Wave proactive rate control. Another important achievement has been the demonstration of ACK train dispersion decorrelation from network latency. This property implies that ACK train dispersion and measured RTT can be considered as two independent input parameters, and as such, they can be fruitfully exploited to discriminate the Bottleneck Capacity (BC) from the Congestion Level (CL). Thus, such measurements are currently used by TCP Wave rate control for the achievement of the optimal operating point, as demonstrated by bottleneck utilization jointly to a stable RTT close to its minimum value, as presented in the paper for a large range of bottleneck capacity values.

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