Cross Layer Interaction for improving the performance of TCP in Multihop Wireless Networks

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ABSTRACT

The objective of the Transmission Control Protocol (TCP) is to provide reliable end-to-end delivery of data over unreliable networks. Over the past few years, the problem of congestion control has received wide-spread attention. Many authors have reported that the TCP interacts with the lower layers, but still it cannot predict route failures and network congestion. Their proposals involving the network layer suggest notifying the TCP sender about a routing failure, when the routing layer detects one. The issues discussed above, with the possibility of further avenues for improving the performance of TCP in multihop wireless networks, served as the motivation for this paper. This paper describes the efficient techniques in various layers to improve the performance of TCP over multihop wireless networks. This work analyses the performance of the proposed two types of cross layer flavors, namely the TCP-AL and TCP-WPAL. The cross layer interaction, TCP-WPAL produced better performance than the TCP-AL.

التفاعل عبر الطبقة لتحسين أداء بروتوكول التحكم بالإرسال (TCP) في الشبكات اللاسلكية متعددة المراحل

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الكلمات الدالة

بروتوكول التحكم بالإرسالTCP، الشبكات اللاسلكية متعددة المراحل ، تحكم الازدحام ، TCP-AL، TCP-WPAL

المُستلخص

يمكن تلخيص الهدف من بروتوكول التحكم بالإرسال (TCP) في إمكانية توفير خدمة توصيل البيانات الموثقة عبر الشبكات غير الموثقة، حيث أن مشكلة التحكم في ازدحام توصيل البيانات قد نالت اهتماماً واسع النطاق خلال السنوات الماضية. و مما يجدر ذكره في هذا المجال توصل كثير من الباحثين إلى أن بروتوكول التحكم بالإرسال تتفاعل مع الطبقات السفلية، إلا أنها لا تتنبأ بفشل الطريق وازدحام الشبكة. كما وأن اقتراحاتهم والمنطوية على طبقة الشبكة تقترح إخطار مرسل بروتوكول التحكم بالإرسال عن فشل التوجيه عندما تقوم طبقة التوجيه بالكشف عن ذلك. وعليه شكلت القضايا والأمور والتي تمت مناقشتها في دراسات هؤلاء الباحثين دافعاً لإعداد هذه الورقة والتي تستهدفُ إلى إمكانية التوصل إلى مزيد من السبل لتحسين أداء بروتوكول التحكم بالإرسال في الشبكة اللاسلكية مُتعددة المراحل. يدرسُ هذا البحث التقنيات الفعالة في طبقات متعددة لتحسين أداء بروتوكول التحكم بالإرسال عبر الشبكات اللاسلكية متعددة المراحل، كما يقوم هذا العمل بتحليل أداء نوعين من الطبقات وهي (TCP-AL) و (TCP-WPAL). أثبتت الدراسة بأن التفاعل عبر الطبقة (TCP-WPAL) قد أنتج أداء حسنا مقارنةً مع الطبقة (TCP-AL).

Introduction

Transmission Control Protocol (TCP) is the major transport protocol utilized in Internet Protocol (IP) networks. It is a transport layer protocol, and a connection-oriented one, which provides end-to-end reliability. It was specifically designed to provide a reliable end to end byte stream over an unreliable network. The TCP is an adaptive transport protocol that controls it's offered load (through adjusting it's window size) according to the available network bandwidth. It additively increases it's congestion window in the absence of congestion, and throttles down it's window when a sign of congestion is detected. In the wired Internet, congestion is identified by packet loss, which results from buffer overflow events at the bottleneck router.

In the case of wired networks, the data loss is only due to congestion, whereas in a wireless network this is not the case. The data loss in the latter may also be due to link failure, route failure, path asymmetry, channel errors, network partition, mobility, nodes in/out of transmission range, hidden and exposed terminal problems, and due to external environmental conditions, such as the weather. Due to this reason, the TCP does not work efficiently in wireless networks (Xiang Chen, *et al*, 2002).

Multihop wireless networks have several characteristics, different from wired networks (Fu, *et al*, 2003). Firstly, in a typical wireless network that uses IEEE 802.11 MAC, packets may

be dropped, due to either buffer overflow or link-layer contention caused by hidden terminals. Such losses directly affect TCP window adaptation. Secondly, a wireless channel is a scarce, shared resource. Improving channel utilization through spatial channel reuse is highly desirable. Multiple nodes that do not interfere with each other should be encouraged to transmit concurrently. The main aspect of the TCP is congestion control. The TCP uses a number of mechanisms to achieve high performance and avoid congestion collapse, when network performance can fall by several orders of magnitude. These mechanisms control the rate of data entering the network, keeping the data flow below a rate that would trigger collapse. They also yield an approximately max-min fair allocation between flows.

Acknowledgments of the data sent, or lack of acknowledgments are used by the senders to infer network conditions between the TCP sender and the receiver (Christina , & Garcia, 1999). Coupled with timers, the TCP senders and receivers can alter the behavior of the flow of data. This is more generally referred to as congestion control and/or network congestion avoidance.

Ignoring the properties of wireless and Ad-hoc Networks, can lead to TCP implementations with poor performance. Concentration has to be given not only to the transport layer but also to all the layers, to improve the performance of the TCP in wireless networks (Xiang Chen, *et al*, 2002). Here, the problem of congestion control over wireless multi-hop networks is considered. Nodes in such networks are radio equipped and communicate by broadcasting over wireless links. Communication paths between nodes which are not in the radio range of each other are established, by intermediate nodes acting as relays to forward data toward the destination. The diverse applications of such networks range from community based roof-top networks to large-scale ad-hoc networks.

In those networks however, packet losses occur more often due to unreliable links than due to congestion. When using the TCP over wireless links, each packet loss on the wireless link results in congestion control measures being invoked at the source. This causes severe performance degradation. If there is any packet loss in wireless networks, then the reason for that has to be found out. If there is congestion, then only a congestion control mechanism has to be applied. The issues discussed above with the possibility of further avenues for congestion control in the TCP served as the motivation and basis for this work.

Material and Methods

(1) Related Work

In this work, the effect of the multihop wireless link on the TCP throughput and loss behavior, and cross layer interaction for several simple network configurations, was studied through analysis; several useful results were obtained. This paper discussed the literatures in the area of the performance of TCP in wireless networks, to improve the performance of the TCP in multihop wireless networks.

(1.1) Transport Layer Solutions

Many authors have suggested several solutions to improve the performance of the TCP in a wireless environment. Some of the solutions are provided in Figure 1.

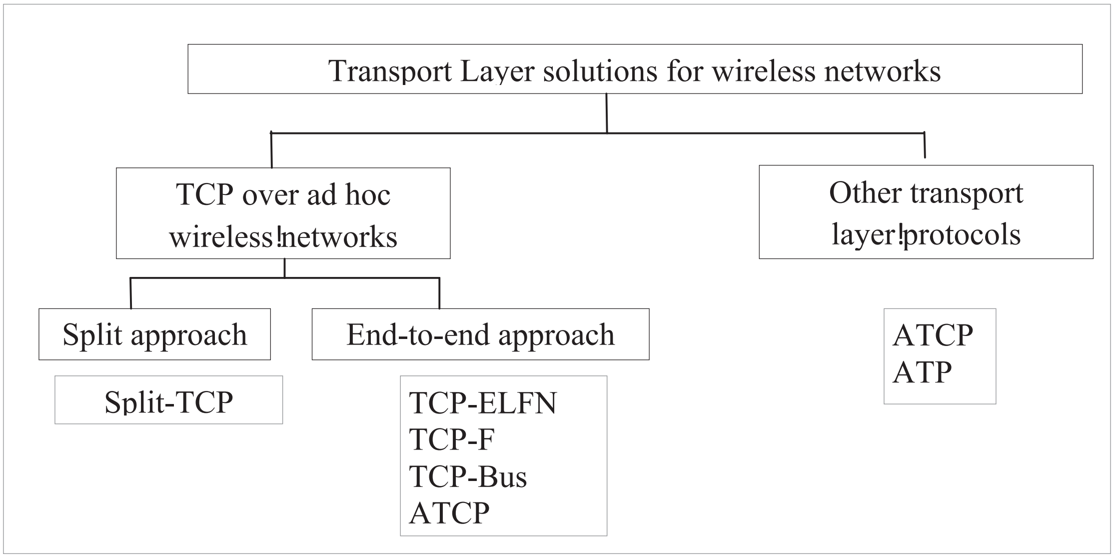

Figure 1: Transport Layer Solutions

(1.1.1) Split Approach

Protocols split each TCP connection between a sender and receiver into two separate connections at the base station- one TCP connection between the sender and the base station, and the other between the base station and the receiver (Balakrishnan, *et al*, 1997). Over the wireless hop, a specialized protocol tuned to the wireless environment may be used.

(1.1.2) Split-TCP

TCP connections that have a large number of hops suffer from frequent route failures due to mobility. To improve the throughput of these connections and to resolve the unfairness problem, the split TCP scheme was introduced to split long TCP connections into shorter localized segments (Balakrishnan, *et al*, 1997). The interfacing node between two localized segments is called a proxy. The routing agent decides if its node has the role of proxy, according to the inter-proxy distance parameter. The proxy intercepts the TCP packets, buffers them, and acknowledges their receipt to the source by sending a local acknowledgement. Also a proxy is responsible for delivering the packets, at an approximate rate, to the next local segment.

(1.1.3) End-To-End Approach

Although a wide variety of TCP versions are used on the Internet, the current de facto standard for TCP implementations is the TCP Reno. They call this the E2E protocol, and use it as the standard basis for performance comparison (Balakrishnan *et al*, 1997). The E2E-NEWRENO protocol improves the performance of the TCP-Reno after multiple packet losses in a window by remaining in the fast recovery mode, if the first new acknowledgment received after a fast retransmission is "partial", i.e., it is less than the value of the last byte transmitted when the fast retransmission was done.

(1.2) Various TCP Versions

(1.2.1) TCP Tahoe

Two variables are used to achieve congestion control in TCP Tahoe (Jacobson 1988). The congestion window (cwnd) governs the amount of data a connection is currently allowed to transmit, while the slow start threshold (ssthresh) determines in which phase of the congestion control procedure the connection is currently in. Tahoe's congestion control mechanism consists of two phases: Slow Start, which is executed at the beginning of the connection and after every packet loss, and

Congestion Avoidance, which is entered into when cwnd reaches the value of ssthresh.

(1.2.2) TCP SACK

The TCP SACK (Mathis, *et al*, 1996) uses an option field to tell the sender which blocks of data have been received. This is to avoid retransmission of packets that have already been received. The advantage of the SACK is seen, when there are multiple dropped packets in a window, because the sending side will know which non-contiguous segments have been received, and thus will not have to be retransmitted. This is in contrast to the situation when SACK is not used; the only information known is the last received acknowledged packet.

(1.2.3) TCP Reno

The TCP Reno (Jeonghoon, *etal*, 1999) advanced the Fast Transmit, where three duplicate acknowledgments signaled a re-transmittance, without a timeout and with Fast Recovery. Fast Recovery meant that once a certain threshold of acknowledgements was received, the window size was decreased by half, rather than starting all again over with slow start. Only during timeout does it go back into slow-start.

(1.2.4) TCP Vegas

The TCP Vegas (Jeonghoon, *et al*, 1999) uses packet delay as an indication of congestion. In a situation when a duplicate acknowledge is received, the timestamp for the acknowledgement is compared to a timeout value. If the timestamp is greater than the timeout value, the Vegas will retransmit rather than wait for three duplicate acknowledgements.

(1.2.5) TCP New Reno

The TCP New Reno (Nadim, *et al*, 2006) responded better compared to the TCP Reno, with the interpretation of partial acknowledgements as indications of packet loss, and not removing it from the Fast Recovery phase. Because the timeout timer is renewed when acknowledgements are received, the New Reno is able to maintain a high throughput.

(1.3) Congestion Avoidance Techniques

(1.3.1) Drop Tail

Dropped packet happens when the buffer queue for a router is overloaded and the last packets to be queued are dropped (Ashish, *et al,* 2012). The router pays no attention to fairness, and has a first-in first-out mentality.

(1.3.2) RED

RED queuing detects when the router is about to hit capacity, and will randomly drop packets to indicate TCP flows, that there is congestion in the network. RED fairly drops packets based on the amount of bandwidth used by a connection, rather than whoever was unfortunately queued at the end of the buffer as in Drop Tail queuing (Ashish , *et al,* 2012). TCP connections detect the lost packet and will back off accordingly.

RED is an active queue management mechanism in routers, which detects congestion before the queue overflows, and provides an indication of this congestion to the end nodes. A RED router signals incipient congestion to the TCP, by dropping packets probabilistically before the queue runs out of buffer space. The RED router operates by maintaining two levels of thresholds: the minimum (min_th) and the maximum (max_th). It drops packets probabilistically if and only if the average queue size lies between the min_th and max_th.

(1.3.3) Explicit Bad State Notification

Explicit Bad State Notification (EBSN) proposed a mechanism to update the TCP timer at the source, to prevent the source form decreasing its congestion window provided there is congestion (Hala Elaarag 2002). The EBSNs are sent to the source after every unsuccessful attempt by the base station to transmit packets over the wireless link. The EBSN would cause the previous timeouts to be cancelled and new timeouts put in place, based on the existing estimate of round trip time and variance. Thus, the new timeout value is identical to the previous one. The EBSN approach does not interfere with the actual round trip time or variance estimates, and at the same time prevents unnecessary timeouts from occurring. This prevents timeouts for packets that had already been put on the network before the wireless link encountered the bad state.

(1.3.4) Explicit Congestion Notification

Explicit Congestion Notification (ECN) (Sally Floyd 1994) is an extension proposed to Random Early Detection (RED). The ECN relies on the extension to the RED, which marks a packet instead of dropping it in when the average queue size is between the min_th and max_th. Upon receipt of the congestion marked packet, the TCP receiver informs the sender (in the subsequent acknowledgement) about the incipient congestion, which in turn, will trigger the congestion avoidance algorithm at the sender. ECN requires support from both the router as well as the end hosts, i.e., the end host TCP stack needs to be modified. If the ECN support is provided then the packets are referred to as ECN capable packets. Packets which are not ECN capable, will continue to be dropped by the RED.

(1.3.5) Multiple Acknowledgements

The multiple acknowledgements (Saad Biaz, *et al,* 1997) method distinguishes the losses due to congestion, or other errors on the wired link, and those on the wireless link. This method uses two types of ACKs to isolate the wireless host and the fixed network: ACKp: This partial acknowledgement with the sequence number Na informs the sender S that the packets(s) with the sequence number up to Na –1 have been received by the base station, and ACKc: This complete acknowledgement has the same semantics as the normal TCP acknowledgement, i.e, the receiver R received the packet.

The sender strictly follows the regular TCP when sending packets with slow start, congestion avoidance, fast retransmit and fast recovery.

(1.3.6) Explicit Loss Notification

Explicit Loss Notification (ELN) (Buchholcz, *et al,* 2005) adds an ELN option to TCP acknowledgements. When a packet is dropped on the wireless networks, future cumulative acknowledgements corresponding to the lost packet are marked, to identify that a non-congestion related loss has occurred. Upon receiving this information along with the duplicate acknowledgements, the sender may perform retransmissions without invoking congestion control procedures.

(2) Enhancements in Data Link Layer

This paper describes the techniques used in the data link layer to improve the performance of the TCP in wireless networks and MANETs. In the data link layer, the Link Random Early Detection (LRED) technique is used to solve the hidden terminal problem, and the Adaptive Pacing (AP) technique is used to solve the exposed terminal problem. This combination of AP and LRED is known as TCP-AL.

(2.1) Link RED

Link Random Early Discard (LRED) is a technique based on queuing methodology. It is based on the observation that the TCP can potentially benefit from the built-in dropping mechanism of the 802.11 MAC. The main idea is to further tune up the wireless link's drop probability, based on the perceived link drops. While the wired RED provides a linearly increasing drop curve as the queue exceeds a minimum value min_th, the LRED does so as the link drop probability exceeds a minimum threshold. In the Data link layer, the LRED technique seeks to react earlier to link overload and solves the hidden terminal problem.

In the LRED, the link layer maintains the average number of the retries for recent packet transmissions. The head-of-line packet is dropped/marked from the buffer with a probability based on this average number. At each node, if the average number of retries is small, say less than min_th, which means that the node is rarely hidden, the packets in the buffer are not dropped/marked. When it gets larger, the dropping/marking probability is computed, and the

minimum value of the computed drop probability and a maximum bound max_p is used. The marking probability is computed as:

$$\text{mark\_prob} = \min \left\{\frac{\text{avg\_retry} - \text{min\_th}}{\text{max th} - \text{min th}}, \text{max\_p}\right\} \quad \textbf{(1)}$$

In the LRED (Fu, *et al,* 2003), the average number of retries is calculated by using the information of the MAC layer; the old average retry value is changed, and the new average retry value is achieved as:

new avg_retry = 7/8 old avg_retry + 1/8 retry **(2)**

When this new avg_retry < min_th, the pacing, extra back off time is provided. This will avoid the loss of packets due to the inherent properties of the wireless networks.

To summarize, the LRED is a simple mechanism, by monitoring a single parameter - the average number of retries in the packet transmissions at the link-layer, accomplishes three goals: a) It helps to improve the TCP throughput, b) It provides the TCP with an early sign of network overload and c) It helps to improve interflow fairness.

(2.2) Adaptive Pacing

The goal of the adaptive pacing technique is to improve spatial channel reuse, by distributing the traffic among intermediate nodes in a more balanced way, while enhancing the coordination of the forwarding nodes along the data path. It solves the exposed terminal problem. This design works in concert with the 802.11 MAC.

In the current 802.11 protocol, a node is constrained from contending for the channel, by a random backoff period plus a single packet transmission time, that is announced by its immediate downstream node. However, the exposed receiver problem persists, due to lack of coordination between nodes that are two hops away from each other. Adaptive pacing solves this problem, without requiring nontrivial modifications to the 802.11 or a second wireless channel. The TCP-AP is to let a node further back-off an additional packet transmission time when necessary, in addition to its current deferral period (i.e., the random backoff, plus one packet transmission time). This extra backoff interval helps in reducing the contention drops caused by exposed receivers, and extends the range of the link-layer coordination from one hop to two hops, along the packet forwarding path.

The algorithm works together with the LRED as follows: Adaptive pacing is enabled by LRED. When a node finds its average number of retries to be less than min_th, it calculates its backoff time as usual. When the average number of retries goes beyond min_th, adaptive pacing is enabled, and the backoff period is increased by an interval equal to the transmission time of the previous data packet. In this way, a better coordination among nodes is achieved under different network load.

The hidden terminal and exposed terminal problems are the main problems of wireless networks. The hidden terminal problem is solved by the use of LRED technique and the exposed terminal problem is solved with the use of AP. These data link layer techniques reduce the congestion and they improve the functionalities of the transport layer by the enhancement of the data link layer.

(3) Cross Layer Interaction

The cross layer interaction techniques used in the lower layers of the protocol stack, solve the hidden and exposed terminal problems of wireless and ad hoc networks. All these techniques used in the lower layers improve not only the lower layer functionalities, but also the TCP congestion control mechanisms in wireless networks. The cross layer interaction i.e. the combination of TCP-AL with IEEE 802.15.4 PHY and 802.15.4 MAC is known to be TCP-WPAL.

(3.1) IEEE 802.15.4 PHY

The IEEE 802.15.4. PHY provides two services: the PHY data service and the PHY management service, interfacing the Physical Layer Management entity (PLME) (Sinem, 2004). The PHY data service enables the transmission and reception of the PHY protocol data units (PPDU) across the physical radio channel. The functions of the PHY are the activation and deactivation of the radio transceiver Energy Detection (ED), Link Quality Indication (LQI), channel selection, Clear Channel Assessment (CCA) and transmitting as well as receiving packets across the physical medium.

The receiver energy detection (ED) measurement is an estimate of the received signal power within the bandwidth of an IEEE 802.15.4 channel. The LQI measurement is a characterization of the strength and/or quality of a received packet. A clear channel assessment (CCA) is performed according to at least one of the following three methods: (a) Energy above threshold; (b) Carrier sense only; & (c) Carrier sense with energy above threshold

(3.2) IEEE 802.15.4 MAC

The IEEE 802.15.4 MAC sublayer provides two services: the MAC data service and the MAC management service interfacing to the MAC sublayer management entity (MLME) service access point (SAP) (MLMESAP). The MAC data service enables the transmission and reception of the MAC protocol data units (MPDU) across the PHY data service) (Sinem, 2004). The features of the MAC sub-layer are beacon management, channel access, GTS management, frame validation, acknowledged frame delivery, association and disassociation.

The IEEE 802.15.4 allows the optional use of a super frame structure, which is shown in Figure 2. The format of the super frame is defined by the coordinator. The super frame is bounded by the network beacons and is divided into 16 equally sized slots. The beacon frame is sent to the first slot of each super frame. If a coordinator does not want to use the super frame structure, it may turn off the beacon transmissions. The beacons are used to synchronize the attached devices, to identify the PAN and to describe the structure of the super frames.

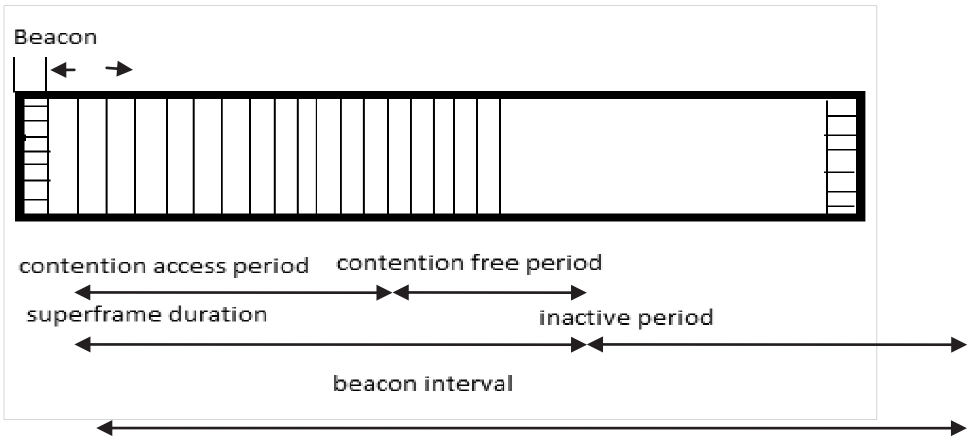

Figure 2: Super-frame Structure

The super frame can have an active and an inactive portion. During the inactive portion, the coordinator shall not interact with its PAN and may enter a low-power mode. The active portion consists of a Contention Access Period (CAP) and a Contention Free Period (CFP). Any device wishing to communicate during the CAP shall compete with other devices using a slotted CSMA/CA mechanism. On the other hand, the CFP contains Guaranteed Time Slots (GTSs). The GTSs always appear at the end of the active super frame, starting at a slot boundary immediately following the CAP. The PAN coordinator may allocate up to seven of these GTSs, and a GTS can occupy more than one slot period.

Cross layer interaction means that the TCP-AP which is concentrating the link layer and, LRED which is the concentrating data link layer techniques, are combined with the IEEE 802.15.4 PHY and IEEE 802.15.4 MAC. Cross layer interaction exploits the dependencies and interactions between layers to increase the performance in certain scenarios of wireless networks. Cross layer interaction does the sharing of knowledge about the layer state and conditions, which are a promising paradigm for performance optimization in wireless systems. It also provides knowledge about the channel conditions of PHY and MAC to routing, transport and application layers, which allow to design more sophisticated allocation and optimization algorithms.

In this work combines the concepts of the TCP-AP, LRED combined with the IEEE 802.15.4 WPAN. Here, the Cross layer interaction is achieved as the TCP-AP which is the concentrating link layer, and LRED which is the concentrating data link layer techniques are combined with the IEEE 802.15.4 PHY and IEEE 802.15.4 MAC. Here, the data link layer, Network layer, MAC layer, Application layer and Physical layer are modified, to improve the performance of the transport layer protocol. Because of the concentration in various interlinked layers improves the performance of the TCP in wireless networks, is improved.

(4) Analytical Results On Tcp-Al And Tcp-Wpal

This work explains about the performance of TCP-AL (TCP with Adaptive Pacing and LRED) and TCP-WPAL (TCP-AL with IEEE 802.15.4 PHY and IEEE 802.15.4 MAC) analytically. These techniques improve the performance of TCP by increasing PDR and Window Size and by decreasing Delay and Jitter. The network parameter computation of TCP-AL and TCP-WPAL are analysed and their improvements are specified.

(4.1) TCP-AL Analysis

In traditional TCP, the average length of packets in queue is described by

avg_len = (1-weight) * avg_len + weight * SampleLen **(3)**

where: avg_len ->average length of queue; weight -> smoothing factor ranges between 0 and 1; and SampleLen -> length of queue.

In general, the delay is given by:

delay = propagation delay + transmit delay + queue delay **(4)**

where:

$$\text{propagation delay} = \frac{\text{Distance between Source and Destination}}{\text{Speed of light}} \quad \textbf{(5)}$$

$$\text{transmit delay} = \frac{\text{Size of the transmitted packet}}{\text{bandwidth}} \quad \textbf{(6)}$$

queue -> packets in queue

In Random Early Detection (RED), the packets are dropped randomly when certain amount of packets (threshold value) are in queue. That is done as shown below.

Begin:
If avg_len ≤ min_th
queue the packet
else if min_th < avg_len
calculate the probability, p
and drop the packet according to p
else max_th ≤ avg_len
drop the packet
end.

where: min_th -> minimum threshold; & max_th -> maximum threshold

In link RED, the probability and dropping of packets are decided not only by the length of queue and additionally it is using the parameters: the average number of retries (avg_retry) and the mark probability (mark_prob). If the avg_retry is less than min_th, mark_prob is zero else it is computed by equation (1).

Begin:
If avg_retry < min_th
mark_prob = 0
else

$$\text{mark\_prob} = \min \left\{ \frac{\text{avg\_retry - min\_bh}}{\text{max\_th - min\_th}}, \text{max\_p} \right\}$$

end.

where: $\text{avg\_retry}_{new} = 7/8\ \text{avg\_retry}_{old} + 1/8$ retry; & max_p -> maximum probability

Normally the relation between dropping of packet and the band width is directly proportional i.e.

drop packet µ share of bandwidth **(7)**

and the bandwidth utilization is given by:

$$\text{bandwidth utilization} = \frac{\text{throughput}}{\text{bandwitdth}} \quad \textbf{(8)}$$

Here if the drop of packets is reduced, then obviously bandwidth will be reduced. Hence the bandwidth utilization is increased. This in turn increases the throughput by the above equation. The avg_retry parameter used in the LRED reduces the drop of packets and increases the throughput. According to the threshold values i.e. min_th and max_th, the drop of packet is decided which in turn reduces the queue length so that the total transfer time reduced. (Nader, 2007) had given the relation between throughput and total transfer time as:

$$\text{Throughput} = \frac{\text{file size}}{\text{Total Transfer Time}} \quad \textbf{(9)}$$

where: file size -> packet size; and total transfer time -> the time taken to transmit a packet

When the total transfer time is reduced, obviously the throughput gets increased. In traditional TCP, the round trip time is calculated with the use of the window size, which is given as:

$R_{TTnew} = \alpha R_{TTold} + (1-\alpha) M$ **(10)**

where α is smoothening factor lies between 0 and 1; and M is the current R_{TT}

If there is a delay and estimated R_{TT} < actual R_{TT}, then lower bound delay is given by:

delay = $2 * R_{TTnew} + O/R$ **(11)**

where: O/R is the time taken by a client to receive the data

Also, the delay is given by:

delay = $2 * R_{TT} + O/R + (k-1)\ [S/R + R_{TT} – WS/R]$ **(12)**

where S/R is the time taken to receive successive acknowledgement, WS is varying window size.

By the Adaptive Pacing (AP) technique, if pacing is ON,

Begin:
extra_Backoff = TX_Timed(DATA) + overhead
backoff = random_Back off + extra_Back off
end

In traditional TCP, the retransmission time out is given by,

$RTO = R_{TT}\ \beta$ **(13)**

where β is the delay variance (Jitter).

By the provisioning of pacing i.e. providing extra back off, the R_{TT} increased. Also the window size fluctuated and increased. Hence in TCP-AL, the retransmission time decreases since, the R_{TT} increases and the RTO decreases and then β reduces. That means the delay and jitter get decreased.

Also, the throughput is given by

$$\text{Expected throughput} = \frac{\text{Congcstion Window}}{\text{Minimum } R_{TT}} \quad \textbf{(14)}$$

Whenever the congestion window is increased, the throughput gets increased. Also when the queue length is reduced, the delay decreases by the equation (4).

(4.2) TCP-WPAL Analysis

In TCP-AL, the standard IEEE 802.11 is used whereas in TCP-WPAL, the standard IEEE 802.15.4 is used. The difference between both standards is shown in Table. 1. IEEE 802.15.4 is the low power, low data rate and low packet size, beacon enabled standard. That is well suited for shorter range of transmission.

Table1: WLAN vs WPAN

Technology	WLAN (IEEE)	WPAN (IEEE)
Standard	802.11 Legacy	Zig Bee 802.15.4-2003 802.15.4-2006
Release Year	1997	2003 and 2006
Frequency Band	2.4GHz	868 MHz, 915 MHz, 2.4 GHz
Maximum Range	~70 meters	~100 meters
Maximum Data Rate	2Mbps	250 Kbps
Number of Users	Dozens	Dozens
Access Method	DSSS, FHSS	DSSS
Modulation Method	GFSK, BPSK, DBPSK, DQPSK	BPSK (868/928MHz) OPSK (2.4GHz)

The difference of parameters between WLAN and WPAN are frequency band, maximum range and maximum data rate only. All other specifications are same. Because of low data rate packets are transmitted in TCP-WAPL technique, the Packet Delivery Ratio (PDR) is increased better and by the use of given equation, throughput is reduced.

$$\text{Throughput} = \frac{\text{file size}}{\text{Tatal Transfer Time}} \quad \textbf{(15)}$$

Even though the total transfer time is minimized, the throughput is reduced because of the lower packet size. Normally the delay is calculated by the equation (4). In TCP-WPAL, the transmission is taking place between shorter distances, propagation delay is reduced and the window size gets decreased so that transmit delay is maintained. Queue delay is also reduced. As all, delay is minimized a lot.

Conclusion

In the data link layer, the enhancement is made by combining the adaptive pacing (AP) and link RED (LRED) techniques has been developed and that is known as TCP-AL. To enhance the performance of TCP with the use of cross layer interaction, the IEEE 802.15.4 PHY and IEEE 802.15.4 MAC are combined with TCP-AL and that is known as TCP-WPAL. The existing

enhancements in the TCP are not suitable for a wireless environment. Also, those mechanisms are concerned about any one of the lower layers and not with all the lower layers. This work suggests the enhancement in data link layer (TCP-AL) and the cross layer interaction, (TCP-WPAL) which provides concentration in the lower layers and to support the functions of the transport layer. This cross layer interaction suits for multihop wireless networks well. Analytically, it is proved that the throughput, PDR in TCP-AL increased and delay, jitter in TCP-AL reduced. Also it proved that the throughput, jitter and delay reduced and the PDR increased in TCP-WPAL. This work can be further extended to find out the threshold - packet size in various scenarios that will produce higher throughput in TCP-WPAL and still lower delay in the TCP-WPAL. Also, it can be applied to test the performance of the TCP in multimedia communication over wireless networks.

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