# Congestion Control with Explicit Feedback for Time Varying Capacity Media

**Sowmya A N Gowda, Bhaskar Rao N**

***Abstract—: XCP is a serious candidate to replace TCP congestion control in the internet. TCP, despite performing end-to-end congestion remarkably well degrades network performance due to unstable throughput, limited fairness and limited fairness. In XCP the routers provides the explicit feedback about the link capacity to the source. In time varying capacity media such as IEEE 802.11 knowing this value is difficult as it depends on many variables. We explore three algorithms for time varying capacity media which maintain efficiency under such conditions. Finally we compare our proposal with TCP new Reno and how such algorithm outperforms in terms of efficiency.***

***Index Terms— Congestion, Wireless Communication, XCP, Blind, ErrorS, MAC.***

## I. INTRODUCTION

Most current internet applications rely on the transmission control protocol (TCP) [1] to deliver data reliably across the network. TCP, despite performing end-to-end congestion remarkably well it degrades network performance due to increased queuing delay, unstable throughput, and limited fairness. **Congestion** describes a situation of extensive resource use when the supply exceeds the capacity. **Congestion control** is a method used for monitoring the process of regulating the total amount of data entering the network so as to keep traffic levels at an acceptable value. Explicit Control protocol is emerging as one potential solution for overcoming limitations inherent to the current TCP algorithm. End-systems use explicit signaling to tell routers about their preferred send rate. In XCP, routers provide multi-bit [8] feedback to sources, which, in turn, adapt throughput more accurately to the path bandwidth with potentially faster convergence times. Such systems, however, require precise knowledge of link capacity for efficient operation. In the presence of variable-capacity media, e.g., 802.11, such information is not entirely obvious or may be difficult to extract. We explore three possible algorithms [10] for XCP which retain efficiency under such conditions by inferring available bandwidth from queue dynamics.

XCP [5] takes a new approach to a congestion control by letting the routers in the network to return explicit feedback to the hosts. XCP is a serious candidate to replace TCP congestion control in the Internet .No change to bulk of TCP kernel code or API semantics .Excellent performance, link utilization, and packet loss across wide ranges of RTT, link bandwidth, flow size .Capable of fair coexistence with other Internet protocols. XCP supports existing TCP semantics, replaces only congestion control, reliability unchanged and no change to application/network API.

Our research explore a design space for explicit congestion protocol (XCP) algorithms which retain efficiency by inferring available bandwidth from queue dynamics. We Test the XCP through simulations with feedback algorithms. Compare the proposals with TCP New Reno and show how such algorithms outperform it in terms of efficiency, stability, queuing delay, and flow rate fairness

## II. RELATED WORK

Our research relies on previous approaches for extending XCP for wireless Capacity media. H. Balakrishnan[1] has proposed a method that analyses congestion control algorithms in context of TCP and other algorithms like RCP, XCP. D. Katabi, M [2] developed a novel approach to internet congestion control that outperforms TCP in conventional environment, and remains efficient, fair, scalable, and stable as bandwidth-delay product increases. The new explicit congestion protocol generalizes explicit congestion notification proposal. K. Ramakrishna [7] specified the incorporation of ECN (Explicit congestion notification) to TCP and IP, including ECN's use of two bits in the IP header. Jianxin Wang [3] proposed a protocol based on VCP and uses end-to-end bandwidth estimation to obtain high resolution congestion estimation. With the estimated available bandwidth and ECN feedback, VCP-BE adjusts the congestion window more precisely than VCP thus converges much faster. Simulation results show that VCP-BE outperforms VCP and MLCP, achieving high efficiency and reasonable fairness. A. Falk [9] specified initial specification for explicit control protocol (XCP), an experimental congestion control protocol .XCP is designed to deliver the highest possible end-to-end throughput over a broad range of network infrastructure. XCP routers are required to perform small calculation on congestion state carried in each data packet. XCP routers also periodically recalculate the local parameters required to provide fairness. [11] Yang Su proposed a wireless explicit control protocol for wireless multi hop network. In our work we present enhancement to WXCP that make it more practical.

## III. EXPLICIT CONTROL PROTOCOL

XCP is a feedback based congestion control system. XCP [4] [6] introduces 20byte header between IP and TCP that carries information about sender's desired bandwidth. It requires all routers and hosts in the network to use the XCP protocol in order to work as intended. Routers in the network provide explicit feedback to the end points advising a change in sending rate [5], [14], [16]

**Manuscript Received on July, 2013.**

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.The end points change their sending rate based on explicit feedback from the network rather than inferring the state of congestion in the network. XCP [15] is novel in separating the efficiency and fairness policies of congestion control, enabling routers to quickly make use of available bandwidth while conservatively managing the allocation of bandwidth to flows. Efficiency controller is responsible for the overall link utilization and fairness controller is responsible for bandwidth [13] distribution among individual flows. The efficiency controller calculates aggregate feedback every control interval using. The EC follows a Multiplicative Increase Multiplicative Decrease (MIMD) rule which achieves a fast convergence to the network conditions and, thus high efficiency. The FC follows an Additive Increase Multiplicative Decrease (AIMD) which is known to converge to fairness resulting in a max-min fair bandwidth distribution among flows.

The EC periodically calculates an amount of bandwidth F to be distributed in the following period. This period is referred to as the control interval, and usually it is set to the average RTT of all flows. The amount of bandwidth F is calculated as:

$$F = \alpha.\ (C - y\ (t)) - \beta.\frac{q}{d} \qquad (1)$$

Where y (t) is the incoming bandwidth, C is the link capacity, q is the length of the persistent queue and d is the control interval duration. The control interval duration is typically set to match the average RTT of the flows traversing this router in order to preserve stability. α and β are system constants dimensioned also to preserve system stability. The persistent

Queue represents the lowest queue length observed within the control interval. As said, F represents the amount of bandwidth that is going to be distributed in the following control interval

### A. Congestion header

Fig. 1 shows XCP Congestion header [12] .Where Protocol Field Indicating next-level protocol used in data payload. Length -Header length in bytes. Version is the version of the congestion header. Format Indicates header format. RTT is Round trip time as measured by sender. X Indicates desired change in throughput by sender. Delta Throughput Inter-packet time of the flow as calculated by sender. Reverse Feedback is Feedback value of Delta Throughput received by the data receiver.

| Protocol | Length | Version | Format | Unused |
|---|---|---|---|---|
| RTT | | | | |
| X | | | | |
| Delta Throughput | | | | |
| Reverse Feedback | | | | |

Fig.1 XCP congestion header

### B. Capacity estimation error

Choosing a capacity value C for the computation of F when the underlying medium allows concurrent access from different stations that can use different rates, e.g., IEEE 802.11, is not trivial. The usable capacity of a medium C, depends on the data rate used by each station, the number of active stations, the number of collisions, failed transmissions, and handshake mechanisms (RTS/CTS) and their thresholds. Previous studies [17] and [4] have shown that XCP is able to compensate an erroneous capacity estimation up to a certain limit by building up the queue. The queue length required to compensate the error is proportional to the error € itself, to the average RTT of the flows d, and the ratio α/β. The error € can be defined as

$$€ = C - C_{real} \qquad (2)$$

Where C is the capacity estimate and Creal represents the actual medium capacity. Substituting (2) into (1), we obtain the aggregate feedback F with capacity estimation error:

$$F = \alpha.\ (C_{real} + € - y(t)\ ) - \beta.q/d \qquad (3)$$

Full utilization of the real link capacity, Creal= y (t), results in an aggregate feedback of zero, since no bandwidth is free to distribute among flows. Under such conditions, we may further simplify (3) and obtain the amount of queue build-up required to compensate a given estimation error.

$$q = \frac{\alpha}{\beta}.\ €\ .d \qquad (4)$$

For a more convenient analysis, we decompose d to reflect the effect that queue build-up has on the overall system delay:

$$d = d_0 + \frac{q}{C_{real}} \qquad (5)$$

Where d0 represents the system base delay that is the system delay excluding queuing delay.

## IV. ALGORITHMS PROPOSED FOR TIME VARYING CAPACITY MEDIA

In this section, we present alternative router functions to calculate the aggregate feedback bandwidth F, which allow XCP algorithms to operate in variable-capacity media. The functions proposed remove the need for the router to be configured with the exact medium capacity and they can also adapt to changing bandwidth conditions over time.

We present three interchangeable algorithms: 1) the Blind algorithm measures spare bandwidth from queue speed, 2) ErrorS uses queue accumulation as an indicator of the capacity estimation error, and the 3) MAC



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algorithm which infers available bandwidth from the idle/busy times at the MAC layer.

Blind and ErrorS are based on a novel approach which uses relative information such as queue length variation or queue accumulation over time. This approach enables an XCP router to control the link utilization simply by monitoring its queue state. The third algorithm presented, referred to as MAC, and is a modified version of the Wireless explicit Control Protocol (WXCP). The MAC approach relies on parameters obtained from the Medium Access Control (MAC) layer such as the transmission and idle time periods. The modifications we introduce to WXCP have the objective of making it simpler and more practical to implement, and also to improve its convergence properties.

### *A. Blind Algorithm*

The Blind variant proposes an alternative method for calculating available bandwidth on a link, based on queue speed**.** This algorithm uses the persistent queue length as feedback for estimating link capacity by calculating spare bandwidth as the variation of the queue length over time. The concept of Blind algorithm is that the available bandwidth can be inferred from queue speed. In fact, the rate at which the queue is drained, or at which it builds up, is a fairly accurate estimate of the difference between incoming bandwidth and the medium capacity within a measurement interval.

In order to measure queue speed, queue should not be empty. We overcome this by stabilizing the queue length above zero, under-utilization is easily identified since small fluctuations are absorbed by the queue. For a value κ at which the queue will stabilize, the aggregate feedback F is given by

$$F= -\alpha.\ \Delta q/d - \beta.\ q\text{-}k/d \qquad (6)$$

Where Δq is the variation of persistent queue within a control interval and Δq/d represents the queue speed within the control interval d. k is the target queue length around which queue measurements should be done. K is responsible for two aspects of the algorithm: 1) drive the system to the point of full link utilization, and 2) stabilizing queue length at some positive value so that queue speed can be measured from variations around that value.

### *B. Error Suppression Algorithm*

We introduce another alternative algorithm, the Error Suppression (ErrorS) algorithm. In contrast to Blind, ErrorS assumes that an estimation of the link capacity exists and, by analyzing queue accumulation over time, the controller is able to infer, and remove, the error present in the capacity estimation. With the knowledge of the approximate error present in the capacity estimation, the ErrorS controller compensates this error. The ErrorS algorithm may be particularly useful when the queue controller is able to estimate, even if not exactly, the medium capacity. It may also be useful, when the medium capacity has a smaller degree of variation, while the arrival and departure of flows is still dynamic. ErrorS aims at being less responsive than Blind, allowing a smoother behavior of the queue.

In Section IIB it was shown that the error present in the capacity estimate is compensated by queue build-up, given the error satisfies $\epsilon < \frac{\beta}{\alpha}.\ C_{real}$. Furthermore, we have established the relation between the error € and the queue length q required to compensate this error (Eq. 4). For the reader's convenience, this relation is presented below:

$$q = \frac{\alpha}{\beta}.\ \epsilon.\ d \qquad (6)$$

Solving in order to the estimation error ϙ we obtain:

$$\epsilon = \frac{\beta}{\alpha}.\ d.\ q \qquad (7)$$

Which tells us that queue length may be used to infer the error present in the capacity estimate. ErrorS takes advantage of this property of the system to measure and eliminate the error of the capacity estimate. It does so by introducing the evaluation of the capacity error in the router aggregate feedback formula:

$$F\ (t) = \alpha.\ [C\text{-}y\ (t)] - \mu.\boldsymbol{\xi}\ (t) - \beta.\ \frac{q(t)}{d} \qquad (8)$$

Where α, μ, β are system constants, C = Creal + ϙ is the router estimate of the medium capacity, and ξ (t) represents an approximation of the capacity error ϙ, affected by the factor α μ. We obtain ξ (t) through the accumulation of the instantaneous estimation of the error:

$$\xi(t) = \frac{\beta}{\alpha}\int_0^t \frac{1}{d^2}\ q(t)dt \qquad (9)$$

This equation tells us that, in each control interval, we sum up the instantaneous estimation of the error (ϙ = β α·d · q) to the overall estimate of the capacity error ξ (t). When ξ (t) = α μ · ϙ the error ϙ will be fully suppressed from the feedback function. The constant parameter μ is used to control the aggressiveness with which the error is suppressed from the system, and it should be dimensioned in conjunction with α, β to ensure system stability. We present a stability analysis of the ErrorS system and provide recommended values of α, β, μ that enable system stability under general conditions. A characteristic of the ErrorS algorithm is that, similarly to the Blind algorithm, it is usable only when queue build-up occurs, that is, when the medium is near full utilization. If the capacity estimation C is higher than the actual link capacity $C_{real}$, that is ϙ > 0, the system will be able to reach to full utilization; however, if for any reason C is below, or falls below the link capacity $C_{real}$, the system will not reach full utilization. For this reason, we adopt the same strategy used by the Blind algorithm: the inclusion of the variable parameter κ (t) which represents the target length of the queue. K (t) will be as low as possible when the bottleneck link is fully utilized, otherwise it will increase up to a maximum value of Qχ.

### *C. The Medium Access Control (MAC) Algorithm*

In the MAC algorithm, the router feedback function is modified as follows:

$$F\ (t) = \alpha.\ S\ (t) - \beta.\frac{q(t)}{d} \qquad (10)$$

Where S (t) is the node estimate of the unused bandwidth, based on the MAC idle and busy periods during the last control interval; it is obtained as follows - during a control interval of duration d the transmission medium is perceived busy for $d_{busy}$ seconds, while in the remainder of the time $d_{idle}$ the medium is perceived as idle. The available bandwidth S (t) is calculated by a node by multiplying the estimate of the medium bandwidth share Bs ( t) assigned to that node, by the fraction of the control interval d during which the transmission medium was idle

$$S\ (t) = B_s\ (t).\frac{d_{idle}}{d} \qquad (11)$$


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Where $d_{idle}$ represents the time interval during which the medium was idle in the last control interval. The medium bandwidth share Bs(t) currently being used by the node is given by:

$$B_s(t) = \frac{out(t)}{d_{busy}} \qquad (12)$$

where out(t) represents the amount of data, in bytes, that the station has transmitted in the last control interval d, and $d_{busy}$ represents the time period during which the medium was perceived as being busy during that same control interval. Bs (t) can be also interpreted as the total bandwidth the station would use if the medium was fully utilized. To ensure that new stations have the opportunity to get its fair share of the medium, even when it is fully utilized ($d_{idle} = 0$), we introduce an additional bandwidth distribution parameter that is used when new stations start transmitting:

$$S(t) = B_s(t).\frac{d_{idle}}{d} + v\ (t) \qquad (13)$$

With ϑ (t) given by:

$$V(t) = \begin{cases} 0 & if\ out(t) > 0 \\ \frac{Q_x}{d} & if\ out(t) \approx 0 \end{cases} \qquad (14)$$

Where Qχ has the same meaning and value as in the Blind algorithm. *V* (t) allows nodes with new flows to use an initial fraction of the bandwidth which, otherwise would be impossible to use if the medium was fully utilized ($d_{idle} = 0 \rightarrow S(t) = 0$). After the initial phase, convergence will be ensured by the fairness properties of the medium access control mechanism, even if at a slow pace.

## V. PERFORMANCE EVALUATION

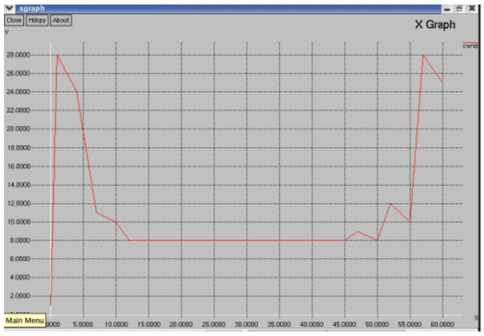
We evaluate the performance of the proposed algorithms-Blind, ErrorS, and MAC by applying to XCP .We compare the proposed algorithms with TCP New Reno.

### A. Simulation setup

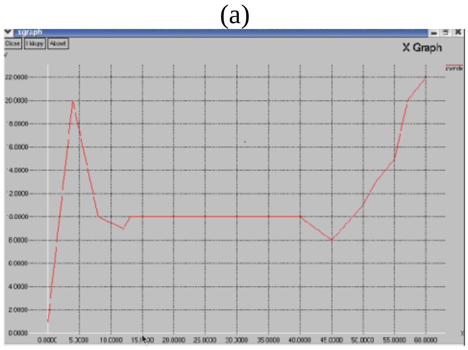
We conduct our simulation using a NS2 simulator. The nodes in the network is randomly distributed. The number of nodes in the network are 50.Our experiment characterizes the dynamics of each algorithm. During the first 20 s, a pair of flows enters the network every 5 s. These 10 flows have a duration of 40 s, which leads to a total experiment duration of 60 s. Propagation RTT is set to 80 ms, neglecting the propagation delay in the wireless hop. This configuration highlights not only how each algorithm performs during stable periods, but also how they react to the arrival and departure of flows. The results of the experiment are plotted in Fig.2, shows the evolution of the congestion window.

### B. Efficiency comparison of proposed with TCP Reno

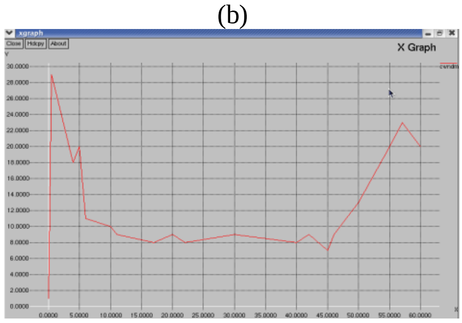
Fig.3 shows how the efficiency scales with increase of the capacity of the wireless medium, normalized to the data rate used in experiment. Reno is always be unable to fully utilize network resources beyond a certain bandwidth threshold. Analyzing the combinations of XCP algorithms, we can conclude that the MAC-based variants are less efficient than Blind and ErrorS.




(a)




(b)




(c)

Fig. 2 Evolution of the congestion window throughput time for all algorithm combination (a) XCP-Blind (b) XCP-ErrorS (c) XCP MAC

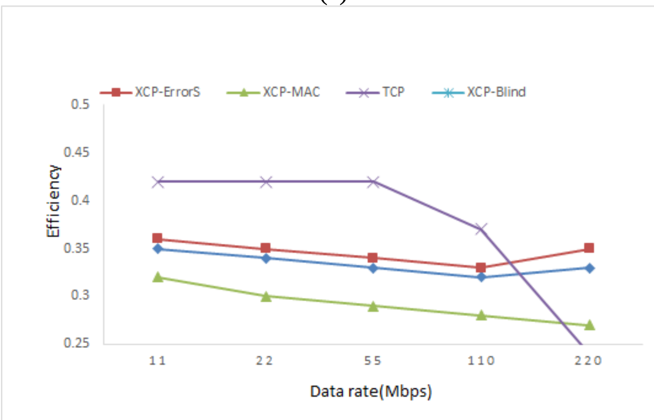



Fig. 3 XCP maintains its properties regardless of the increase of bandwidth while TCP Reno utilization decreases with increase of the medium bandwidth

## VI. CONCLUSION

In this paper, we proposed three alternative algorithms for XCP to work in time varying capacity media: Blind, ErrorS, MAC, which we evaluated through simulation. Blind and ErrorS uses queue speed to infer the capacity of media.MAC algorithm uses information from the MAC



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layer. Simulation results shows that the proposed algorithm maintains most of the properties of XCP and is suitable for wireless time varying capacity media.

## VII. ENHANCEMENT

XCP use explicit feedback to guide endpoint transmission rates for near optimal capacity utilization and fairness. The non-co-operative end hosts can manipulate and ignore feedback

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