Bandwidth Management for Mobile Media Delivery

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***Abstract*—Mobile broadband networks using 3G and post-3G technologies (such as EV-DO, HSPA, WiMAX, LTE) are rapidly becoming one of the prominent means to access the Internet. Multimedia consumption — requiring low delay, high bandwidth, or a combination of both — is projected to become a large portion of bandwidth utilization in mobile broadband networks. In this paper, we study the fundamental problem of how packet loss and delay vary as a function of the transmission rate over these networks. With extensive real-world measurement studies, we analyze the performance of a number of rate control algorithms commonly used in media transmission. We show that the variable nature of congestion signals (loss and delay) on these networks leads to an ultimate failure of existing rate control strategies to deliver adequate performance for multimedia applications. In addition, we show how a rate control algorithm derived from the utility maximization framework — which uses queuing delay as the primary congestion signal — can be modified to solve the challenging issues we have observed. By using a variable threshold to define when the network is congested, our proposed solution is able to achieve a significant improvement over algorithms that use fixed definitions of congestion.**

I. Introduction

Mobile broadband Internet usage is rapidly on the rise. In US and Europe, the proportion of mobile users that access the Internet regularly has quadrupled and tripled over the past four years, respectively [5]. Owners of smartphones are driving the increase in mobile Internet usage, and most consumers access the Web on their smartphones primarily via their cellular connection (rather than Wi-Fi). As a result, 3G cellular technology drives the growth of mobile broadband usage, and is currently available to more than 20 percent of cellular users around the world. 4G mobile broadband technologies, such as WiMAX and LTE, are being quickly deployed as well.

Among smartphone activities, more and more users are utilizing bandwidth intensive applications. Mobile video is experiencing explosive growth and is driving the growth of mobile broadband. According to [20], mobile video has represented 47% of peak hour traffic in November 2010, up from 27% in January 2010. In addition, interactive media applications — such as video conferencing on mobile broadband — are quickly becoming popular. *FaceTime*, introduced by iPhone 4 and further adopted by iPad 2, is redefining the mobile video conferencing experience. By October 2010, 19 million FaceTime-enabled devices had been shipped.

For mobile broadband applications such as playing online games, viewing video streams, browsing maps, or video conferencing, bandwidth plays a critical role in the users' experience, and can mean the difference between a satisfying one vs. a frustrating one. Currently, mobile broadband users are frequently experiencing highly fluctuating bandwidth.

Depending on the interaction between mobile devices and the cloud, there are primarily two categories of mobile broadband applications: *streaming* and *interactive*. For streaming applications, such as map browsing and video-on-demand (e.g., YouTube, NetFlix, Hulu), the effective throughput determines the quality of experience. Other network parameters, such as network latency, jitter (queuing delay), and packet loss, are less relevant. These applications either have a typical response time of several seconds (in map browsing), or build a buffer of several seconds (in video-on-demand apps) at the mobile devices to absorb the jitter and packet loss. On the other hand, interactive applications — such as games and video conferencing — strive to achieve a good throughput with low latency, jitter and packet loss.

As we will discuss in Sec. II, there exists an extensive body of literature on rate and congestion control in the Internet. However, few addresses the unique issue of bandwidth management on mobile broadband networks. Typical rate control algorithms, such as the TCP variants, TCP Friendly Rate Control (TFRC), and other utility maximization approaches, attempt to maximize network throughput and ensure some sense of fairness (to maximize network utility). Rate control is performed by observing congestion signals, which consist of packet loss and jitter, and then adjusting the rate as a function of the congestion level being observed. This works well on the wired Internet when packet loss and jitter accurately reflects congestion. However, for mobile broadband networks, large variations in jitter and loss are observed *even when the channel is not congested*. As an example, 3G and WiMAX networks can experience packet losses of 3-5% and jitter of 50-200 milliseconds, regardless of the transmission rate. Reducing the transmission bit rate does not result in a reduction of the packet loss rate or jitter level, since the existence of some packet loss and jitter does not equate to congestion.

In this paper, we study how packet loss and jitter change as a function of transmission rates in mobile broadband networks. Through extensive experimental results, we show that even at low transmission rates, there is a significant amount of packet loss and jitter, which cannot be attributed to congestion but which is likely due to short-term fluctuations in capacity caused by interference. Using our experimental data, we also show that existing rate control algorithms, which use a fixed *definition of congestion*, perform relatively poorly in terms of throughput and/or packet loss and jitter. By "definition of congestion," we mean jitter or packet loss boundary or threshold between the *congestion* and *congestion-free* zone.

To address this challenge, our original contribution in this paper is inspired by a queuing delay (jitter) based rate control algorithm derived from the utility maximization framework. We improve the algorithm to use a *variable* boundary or threshold between the *congestion* and *congestion-free* zone. The *congestion* zone should be one where reducing the transmission rate results in a noticeable reduction in the congestion level in terms of jitter and/or packet loss. The *congestion-free* zone is one where reducing the transmission rate does not result in a noticeable reduction in the congestion level. We show that by using a *variable* threshold — which is different for each network and learned from network measurements — we can achieve significantly improved performance in terms of throughput, jitter and packet loss. This allows for full wireless link utilization, which is critically important for bandwidth-intensive multimedia applications such as HD playback, while maintaining a low jitter and packet loss, useful to real-time multimedia applications such as conferencing.

The remainder of this paper is organized as follows. In Sec. II, we examine related work on the topic of bandwidth management. In Sec. III, we study an extensive set of representative network traces collected from real network conditions. In Sec. IV we show that existing rate control protocols may perform very poorly in terms of link utilization and/or packet delay and loss. In Sec. V, we present how using variable definitions of congestion can improve the performance of existing rate control algorithms over mobile broadband networks. Sec. VI concludes the paper.

Although we present results for a couple of representative mobile broadband networks — specifically for a 3G EV-DO network (referred to as simply 3G) and for a WiMAX network – we find that the contributions of this paper hold in most other mobile broadband networks as well.

II. Related Work

There exist two major categories of congestion control and rate control algorithms in the literature: (1) those based on the estimation of available bandwidth, and (2) those based on end-to-end congestion control. An example of rate control based on available bandwidth estimation can be found in [21]. The scheme assumes a single critical link along the path. It sends probe packets on the link, estimates the capacity and the cross traffic from the gaps of the incoming and outgoing packet pairs, and calculates the available bandwidth on the link. With one dominant critical link, the scheme can give a quick estimate of the available bandwidth by solving an equation that relates the available bandwidth to the packet gaps received. However, it has been shown [13] that this category of technologies frequently fail in a complex Internet environment, and will completely break down in mobile broadband. Moreover, applications with available bandwidth based rate control will usually fail to share bandwidth fairly.

End-to-end congestion control, such as the standard Additive-Increase Multiplicative-Decrease (AIMD) algorithm used in TCP [11], dominates the Internet. TCP uses a congestion window to control the sending rate which it adjusts using packet loss and/or increased queuing delay as the congestion signal. During the congestion avoidance phase, TCP decreases its window (usually multiplicatively) in response to congestion, and increases its window (usually additively) when no congestion is detected. There has been significant work on TCP congestion control, covering various components. For example, the initial window size could be 1, 2, 3, 4, or even 10 packets; the slow start algorithm could be the standard slow start, limited slow start, hybrid slow start; the congestion avoidance algorithm could be AIMD [11], CUBIC [9], CTCP [22]; the loss recovery mechanism could be Reno and NewReno. In applications such as video conferencing where it is desirable to use a rate control algorithm instead of congestion control, TCP Friendly Rate Control (TFRC) [10] has been developed to achieve rate fairness with TCP, but with a relatively smooth sending rate. Many existing TCP congestion control variants and TFRC interpret any amount of packet loss and increased queuing delay as congestion. Thus, for a noisy channel such as mobile broadband, TCP and TFRC may overreact to the inherent packet loss and jitter and under-utilize the network.

There have been attempts to improve the performance of congestion control protocols such as TCP over networks with wireless links, where not all loss is caused by congestion. A good summary of existing work is provided in [2]. However, existing work has mostly focused on *loss-based* protocols, with the goal being to either hide loss from the upper layer (such as TCP) via retransmissions at the lower layer [1], [3], or modify the upper layer protocol to determine whether observed loss is actually due to congestion [4], [16], [18], [22]. The former approach has been difficult to implement as it requires changes in the hardware or in the firmware which runs the lower layer protocols. The latter approach involves determining what contribution of loss is actually due to congestion. This can be aided by hints from the lower layer, which also requires modifications to the lower layer, or by using additional signals such as jitter (queuing delay). These additional signals can either directly be used in determining congestion or in aiding the rate control algorithm in other ways. From a practical implementation point of view, most of what is currently used is the use of secondary signals to determine when loss is due to congestion. As an example, TCP Westwood [18] attempts to determine which loss is due to congestion by utilizing bandwidth estimation techniques to set the slow start threshold and initial congestion window. This works well when only *loss* is a noisy congestion signal. However, on mobile broadband networks, where *queuing delay* is also a noisy congestion signal, bandwidth estimation techniques do not work well.

Network utility maximization has attracted significant attention since the seminal framework was introduced [15], [17]. In the framework, network protocols are understood as distributed algorithms that maximize aggregate user utility under network resource constraints. The user's utility function is typically assumed to be a strictly concave function of user rate and the resource constraints set is linear. This framework not only provides a powerful tool to reverse engineer existing protocols such as TCP [14], but can be used to develop other network protocols.

In this paper, we revisit this fundamental challenge of determining when observed congestion signals — such as jitter and packet loss — are in fact due to congestion, but over

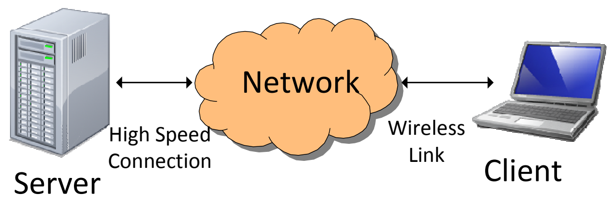


Fig. 1. The system setup for network trace collection. The server is connected to the Internet via a high-speed connection. The client is connected to the Internet via a mobile broadband connection such as 3G or WiMAX.

mobile broadband networks where these signals are inherently *noisy*. However, our work differs along several directions with respect to previous work. First, we deal with congestion control protocols which also use *queuing delay or jitter* as primary congestion signals as opposed to just *packet loss*. Second, we attempt to deal with networks which have *variable levels* of noise in jitter and loss as opposed to *fixed levels*. As additional contributions, we show that mobile broadband networks have very high inherent jitter and packet loss levels, as opposed to traditional networks. This occurs even in the absence of congestion. This requires a *variable* threshold in determining when jitter or packet loss is caused by congestion, which has not been proposed in the literature. Although using a high fixed threshold may result in full link utilization for most networks, it will have a detrimental effect on real-time applications where delay is also important.

III. Empirical Observations

A. Network Trace Collection

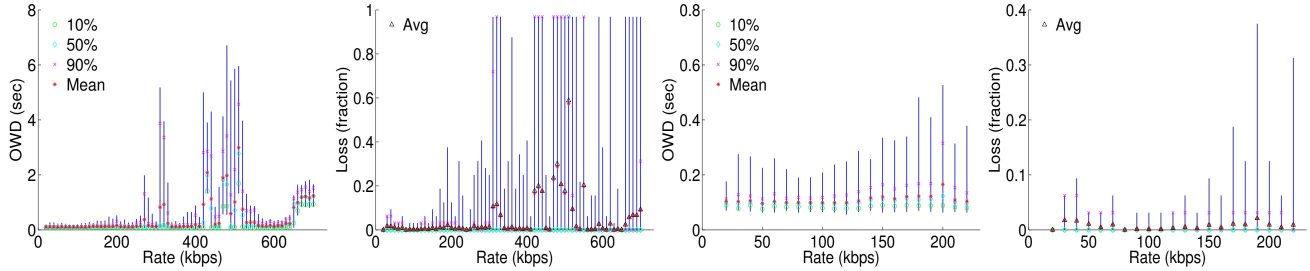
In this section, we show results from extensive network traces that we have collected from 3G and WiMAX networks. Our network trace collection is performed using the setup as shown in Fig. 1, where the server is connected to the Internet via a high-speed Internet link, and the client is connected via a mobile broadband link such as 3G or WiMAX. In such a setup, the most likely place for the bottleneck link is the mobile broadband link. We refer to "upload" traffic when the client sends data to the server and "download" traffic when the server is sending to the client. We send packets in the upload or download direction using a payload size of M bytes at a rate of R bytes/sec. The receiving end observes the arriving packets and records queuing delay and loss measurements. The rate R is varied over various transmission rates, and packets are sent at each rate for a duration of 20 seconds. We have collected traces in both download and upload directions and at different times of the day as well as for various packet sizes.

We believe that these constant bitrate "upload"/"download" settings are the most representative settings in media delivery and media conferencing applications in commercial wireless broadband networks. For example, if we are doing video conferencing at 600 kbps at 30 frames/sec using constant bitrate video coding, each frame is coded to 2500 bytes. It can be packetized to two 1250 byte packets or four 625 byte packets. Note that it is possible to avoid the burst of bitrates caused by periodic insertion of an **I** frame by instead using periodic intra-macroblocks [23]. This strategy can improve error resiliency in addition to making the video conferencing bandwidth smooth. For variable bitrate media delivery, the client typically has a buffer, so assuming a constant bitrate for transmission is still valid.

In the following figures, we show the queuing delay and loss rate observed when sending packets at various rates. The x-axis shows the rate in kbps and the y-axis shows the one-way queuing delay (jitter) (OWD) in seconds for the delay plots, and the loss rate as a fraction for the loss plots. The OWD is estimated as one-half the RTT, although using clock drift compensation techniques, true OWD values can also be obtained. At each particular transmission rate, we draw a straight vertical line, with the bottom of the line representing the minimum queuing delay observed and the top of the line representing the maximum queuing delay observed at that rate. We use a circle, a diamond, and a cross to represent the 10-, 50- (median), and 90-percentile queuing delay observed, respectively, and use a star to represent the average queuing delay at each rate. We visualize the loss data in a similar fashion, except each loss data point is calculated from the packet loss rate in a sliding window of 32 packets. Thus, if we have sent $1,000$ packets at a particular rate, we will have $1000 - 32 = 968$ loss data points. We show representative traces showing the network characteristics in Fig. 2 and Fig. 3, for the 3G and WiMAX networks respectively. These traces are for the upload direction, with a packet size of $P = 1000$ bytes. As a benchmark comparison, we show the network characteristic of a cable modem link in Fig. 5. In these figures, we also show enlarged versions of the queuing delay and loss rate for the rates in the congestion-free zone.

We first observe that for each network (3G, WiMAX, and cable modem), there exists a clear distinction between the *congestion-free* zone and the *congestion* zone. That is, there exists a rate above which the loss and queuing delay statistics clearly start to rise. Thus, there exists a metric, which could be loss, queuing delay, or some combination, that can be used to detect congestion. The 3G and WiMAX networks exhibit much more variation in terms of queuing delay and loss in the non-congested zone than the cable modem link. For the cable modem link, any queuing delay above a small amount (such as 50 ms) can be considered congestion and any loss can be considered congestion as well. For 3G and WiMAX networks, significant variations of loss and queuing delay may be observed even when the network is in the congestion-free zone. For example, we see 2-3% packet loss even in the congestion-free case, along with 50-100 ms queuing delays.

In Fig. 4, we show delay and loss rate statistics for the WiMAX network in the download direction. Although the overall bandwidth is higher in this direction, the queuing delay and loss characteristics are similar to that in the upload direction. In Fig. 6, we show results for the WiMAX network in the upload direction when the packet size is reduced to 50 bytes. Again, we see that although the overall bandwidth decreases, the congestion signals are still fairly noisy. In Fig. 7, we show the results for the 3G network when we collect the trace at night, with fewer users using the network. We see that the overall bandwidth is higher than when operating during the day. However, the inherent noise in the congestion signal is still evidently present, and we still see a clear separation between the congestion-free and congestion zones.



(a) (b) (c) (d)

Fig. 2. Results from network traces showing (a) one-way queuing delay (OWD) and (b) loss rate for the 3G network in upload direction as a function of transmission rate. In (c), we show enlarged version of the OWD results in the congestion-free zone, and in (d), we show the loss rate in the congestion-free zone. We show the range of values observed, with the 10%, 50%, 90% and mean marked. The loss rate shown is computed over a sliding window of 32 packets. The overall average loss rate at a particular bitrate is also marked.

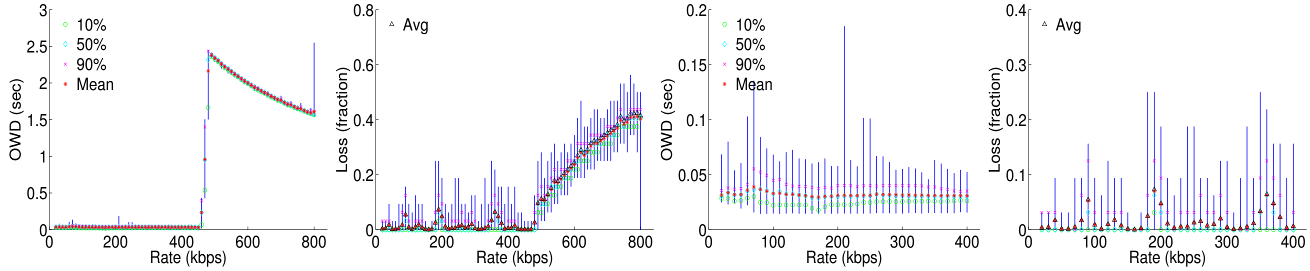


Fig. 3. OWD and loss rate statistics for a WiMAX network trace in the upload direction.

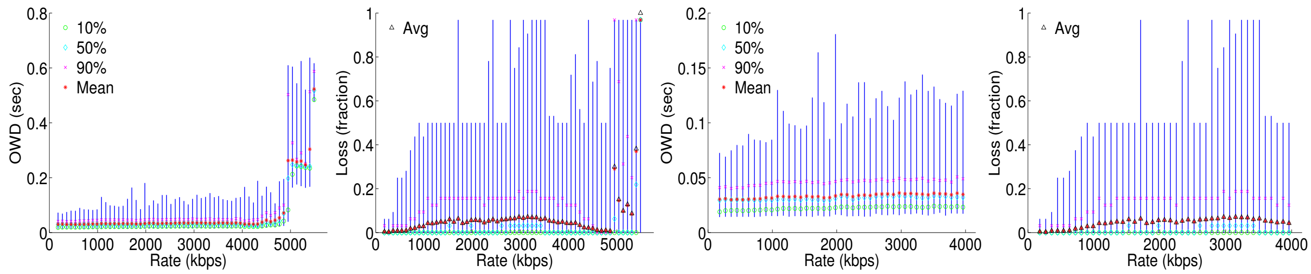


Fig. 4. OWD and loss rate statistics for a WiMAX network trace in the download direction.

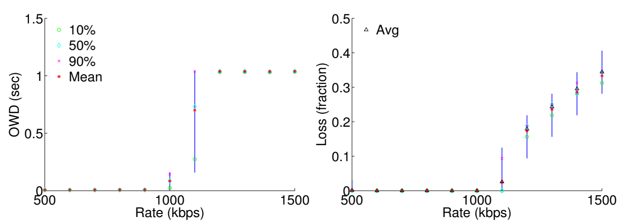


Fig. 5. Statistics for a cable modem connection in upload direction.

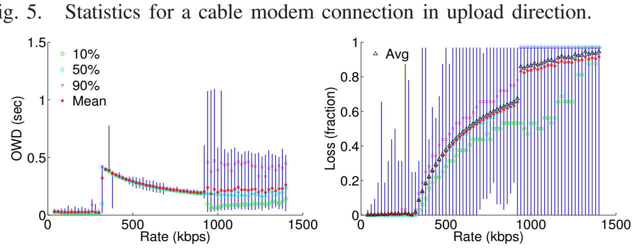


Fig. 6. Statistics for WiMAX network in upload direction with 50B packets.

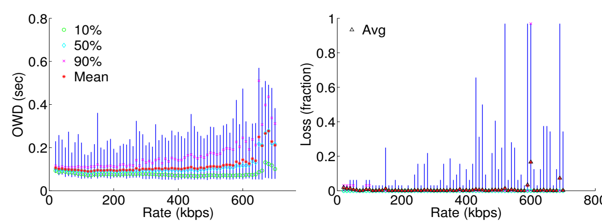


Fig. 7. Statistics for 3G network in upload direction at night.

B. Cause of Noise in the Congestion Signal

The fundamental reason for the inherent and variable nature of jitter and loss in broadband networks is due to rapid and short-term fluctuations in short-term capacity.

We illustrate the problem with a simple example. Consider a mobile broadband network where the mobile link attempts to compensate for short-term fluctuations in the network capacity by performing link layer retransmissions up to 50 times. Suppose the bottleneck link has capacity 500kbps, and the packet size 512 bytes (4096 bits). Then, we may see up to 410ms additional delay (due to the 50 retransmissions) even when the sending rate is below the *average* capacity of the link due to the short-term fluctuations in capacity. Suppose that even after the link layer retransmission, some packets are still lost, resulting in an average loss rate of 0.5% regardless of the transmission rate.

In this case, many rate control protocols would result in link under-utilization. For example, a delay-based protocol may misinterpret the additional delay caused by link layer retransmissions as queuing delay. If such a protocol has an operating point that declares any queuing delay above

TABLE I
RESULTS FROM VARIOUS ALGORITHMS ON 3G NETWORK UPLOAD – "ADAPTIVE" REFERS TO THE ALGORITHM IN SEC. V.

	Throughput (kbps)	Delay (sec)	Loss
NewReno-like	164	0.12	0.041
Vegas-like	209	0.13	0.007
CTCP-like	223	0.15	0.478
TFRC	198	0.10	0.104
UM	219	0.45	0.744
Adaptive	207	0.10	0.062

TABLE II
RESULTS FROM VARIOUS ALGORITHMS ON WIMAX NETWORK DOWNLOAD.

	Throughput (kbps)	Delay (sec)	Loss
NewReno-like	585	0.018	0.019
Vegas-like	578	0.017	0.012
CTCP-like	2629	0.085	0.097
TFRC	1154	0.017	0.044
UM	2285	0.020	0.470
Adaptive	3901	0.042	0.059

200ms as congestion, then this misinterpretation of link layer retransmissions as congestion would prematurely reduce the rate. Similarly, a loss based protocol which declares any loss to be congestion would also prematurely reduce the rate due to the 0.5% inherent loss rate. Such link under-utilization caused by the noise in congestion signals on mobile broadband networks degrades the performance for bandwidth-intensive multimedia applications. Arbitrarily choosing a higher threshold for declaring congestion may result in higher queuing delays and congestion induced packet loss.

IV. RATE CONTROL FOR MOBILE BROADBAND

In this section, we analyze the performance of media rate control algorithms designed from the principles of existing congestion control protocols. We evaluate the following protocols: loss based rate control (TCP NewReno-like) [8], delay based rate control (TCP Vegas-like) [4], loss and delay based rate control (Compound TCP (CTCP)-like) [22], TCP friendly rate control (TFRC) [7], and primal-dual utility maximization (UM) based techniques with fixed parameters [6]. We don't consider available bandwidth estimation techniques [12], as they provide no guarantees of fairness across multiple flows and are known to perform poorly on noisy networks.

Using the collected traces from Sec. III, for the 3G and WiMAX network, we simulate the performance of these rate control protocols. Simulation is used for a fair comparison as the performance of mobile broadband networks changes rapidly over time. For each protocol, we compute the overall transmission rate (throughput), queuing delay, and loss rate. The results are summarized in Table I and Table II. The throughput reported is the rate which is achieved accounting for packet loss ("goodput"). For brevity, results are shown for the 3G network in the upload direction and for the WiMAX network in the download direction. Other configurations — such as 3G in the download direction — show similar results. In the figures for each result, we show the transmission rate along with the PDF of queuing delay and loss rate, using a 32 packet sliding window.

A. TCP-NewReno-like Rate Control

In TCP NewReno-like rate control, we use a method inspired from classical TCP NewReno congestion control [8]. In the simplest form, TCP increments the window by $\frac{M^2}{W}$ for every ACK received and decrements the window by $-\frac{W}{2}$ for every NACK. Since we consider rate adjustment for media applications, we translate the window to transmission rate using $W = R \cdot SRTT$, where $SRTT$ is the smoothed transmission rate. Therefore, we increment the rate by $\frac{M^2}{RSRTT^2}$ for every ACK, and decrement the rate by $-\frac{R}{2}$ for every NACK. The rate change curve vs. congestion level is shown in Fig. 8(a).

When using TCP NewReno-like rate control, we expect that for most networks it results in full network utilization, but at large queuing delays and some amount of congestion induced packet loss. However, for mobile broadband networks, we actually see significant link underutilization. Fig. 9 and Fig. 10 show the results. From Table I, we see that the 3G link is only about 50% utilized and the WiMAX link is only about 12% utilized. This link under-utilization is caused by the fact that in these networks, there is significant random loss, and thus a single packet loss by itself is not necessarily congestion as loss based TCP assumes. From, Fig. 19, we see that loss is not a good signal for congestion detection (especially a single loss event) for mobile networks, and thus a loss based rate control algorithm will not work well. Modifying the definition of congestion to mean a loss rate greater than 20% would improve utilization for the WiMAX network but the operating loss rate would be high.

B. TCP Vegas-like Rate Control

Since loss is not a good congestion signal, we consider a delay based rate control algorithm, similar in spirit to TCP Vegas [4], which uses queuing delay above some threshold as a signal of congestion. In TCP Vegas-like rate control, for each ACK we increase the rate by $\frac{M^2}{R \cdot SRTT^2}$ if $\delta < \kappa$, do nothing if $\kappa \leq \delta < \zeta$, and decrease the rate by $\frac{M^2}{R \cdot SRTT^2}$ if $\delta \geq \zeta$. Upon NACK, we decrease the rate by $-\frac{R}{2}$. The rate change curve is shown in Fig. 8(b).

Fig. 11 and Fig. 12 show the same results as before using the TCP Vegas-like algorithm. From Table I, we see better link utilization than the TCP NewReno-like rate control over a 3G network. However, from Table II, we still see link under-utilization for the WiMAX network in the download direction. Although delay may be a good signal of congestion, the use of fixed parameters makes it suitable for only one type of network. For example, if we optimize the parameters to work well in a 3G network in the upload direction at a particular bitrate, it may sill not work effectively in the WiMAX network in the download direction (still results in link under-utilization) because of the operating point not being correct.

Rate control algorithms with fixed parameters have an operating congestion point which decreases with bitrate. Since the WiMAX network in the download direction has much higher bitrate than the 3G network in the upload direction, the delay operating point is much lower for the WiMAX network in the download direction. Since it is within the inherent noise in delay for this network, it causes link under-utilization.

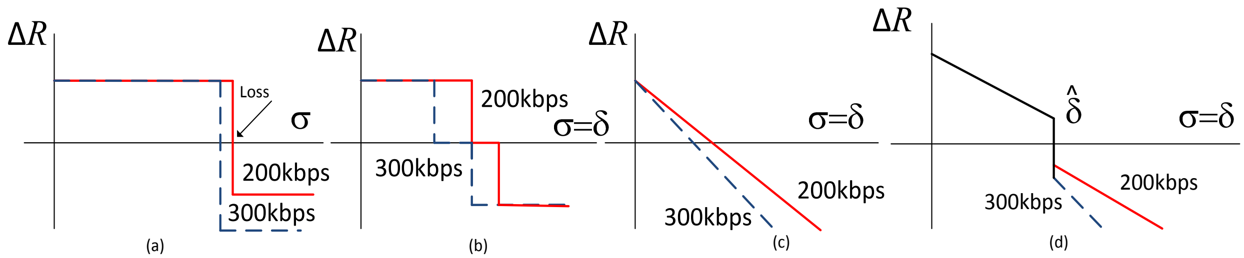
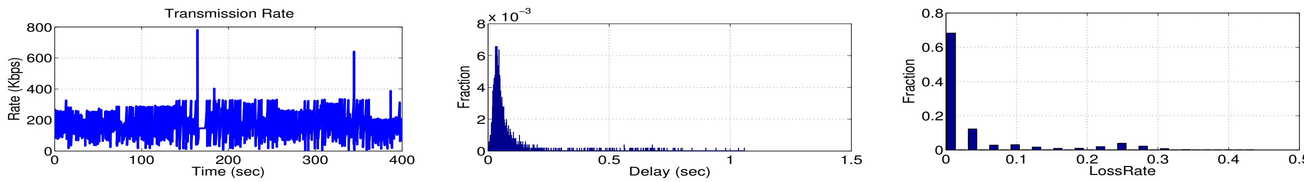
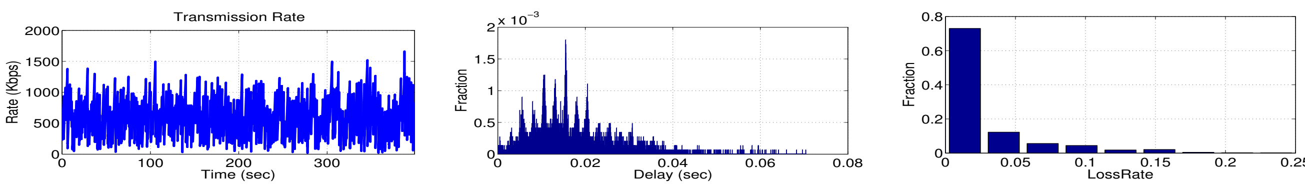


Fig. 8. ΔR vs. ρ for various congestion control algorithms for two different rates, (a) TCP NewReno-like rate control, (b) TCP Vegas-like rate control, (c) Primal-Dual Utility Maximization (UM), (d) Our solution presented in Sec. V. We see that the higher rate curve is always below the lower rate curve in order to guarantee fairness. Both (c) and (d) can easily be modified to have variable congestion thresholds using our proposed strategies from Sec. V-A.



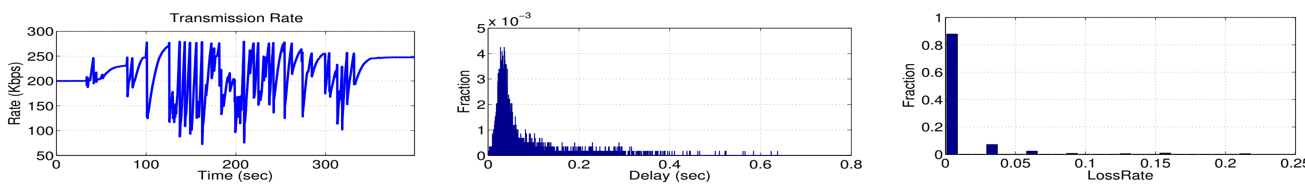
(a) (b) (c)

Fig. 9. (a) Transmission rate, (b) PDF of queuing delay, and (c) PDF of loss rate when using TCP NewReno-like rate control on 3G network.



(a) (b) (c)

Fig. 10. (a) Transmission rate, (b) PDF of queuing delay, and (c) PDF of loss rate when using TCP NewReno-like rate control on WiMAX network.



(a) (b) (c)

Fig. 11. (a) Transmission rate, (b) PDF of queuing delay, and (c) PDF of loss rate when using TCP Vegas-like rate control on 3G network.

C. Compound TCP-like Rate Control

We also evaluate a rate control inspired by a recent TCP variant, Compound TCP [22], which was recently developed for use on high bandwidth-delay product networks. Since it uses both delay and loss signals, a CTCP-like rate control may work better. Fig. 13 and Fig. 14 show the results. Although we see that the link is fairly well utilized in both cases, the use of fixed parameters results in too high of an operating congestion point on the 3G network (over 40% loss), which is unacceptable for real-time applications.

D. TFRC

TFRC [7] exhibits the same throughput, delay, and loss as TCP NewReno-like rate control, with the difference that it has a smoother rate. It still suffers from the issue of link under-utilization. Fig. 15 and Fig. 16 show that the throughput is very similar to TCP NewReno-like rate control. This is understandable as TFRC uses the TCP throughput equation to set the transmission rate [7] and is designed to have similar *average* throughput as TCP NewReno.

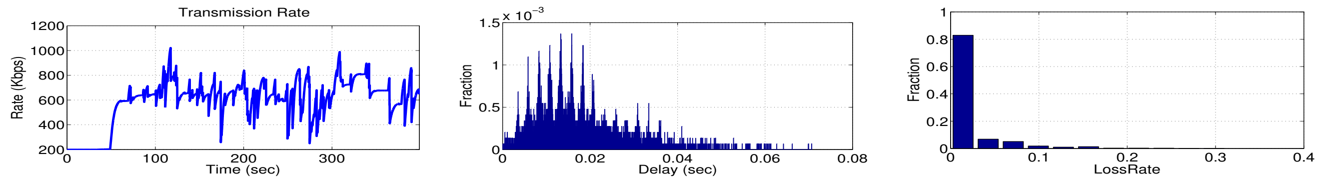
E. Primal Dual Utility Maximization

In primal-dual utility maximization using the log utility function, we attempt to maximize the total utility over the network, in which each source has a utility vs. rate given by $U(R) = k_0 \log(R)$, where R is the rate and k_0 is a constant. Provided the buffers along the path are of a sufficient size (larger than the operating congestion point defined below), this gives a rate control algorithm that adjusts the rate using

$$\Delta R = k_2(k_0 - \delta R)\Delta T, \quad (1)$$

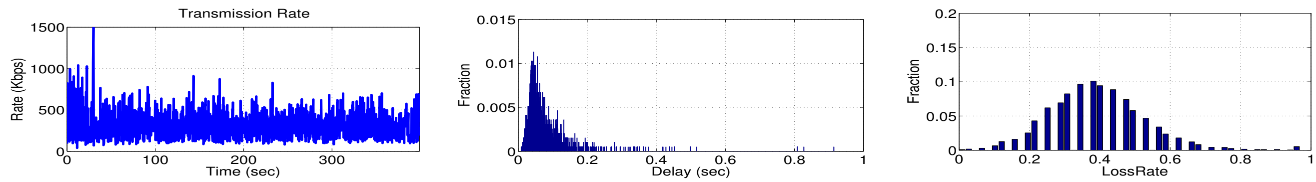
where δ is the observed queuing delay, k_2 is a constant, and ΔT is the time since the previous adjustment.

The operating congestion point is given by δ, where $\Delta R = 0$. This gives an operating congestion point which is a queuing delay of $\hat{\delta} = \frac{k_0}{R}$, where R is the steady state rate. If we fix k_0, then we see that $\hat{\delta}$ is a function of rate. For networks where delay is a noisy signal, if $\hat{\delta}$ is set too low (for example if k_0 is chosen too low), then a delay value may be erroneously classified as congestion and the link will be under-utilized. This would result in poor performance for bandwidth



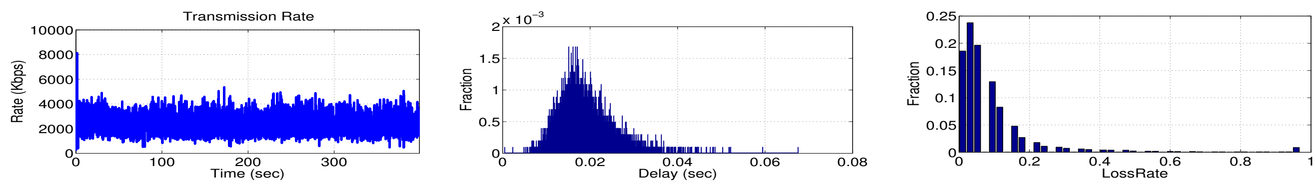
(a) (b) (c)

Fig. 12. (a) Transmission rate, (b) PDF of queuing delay, and (c) PDF of loss rate when using TCP Vegas-like rate control on WiMAX network.



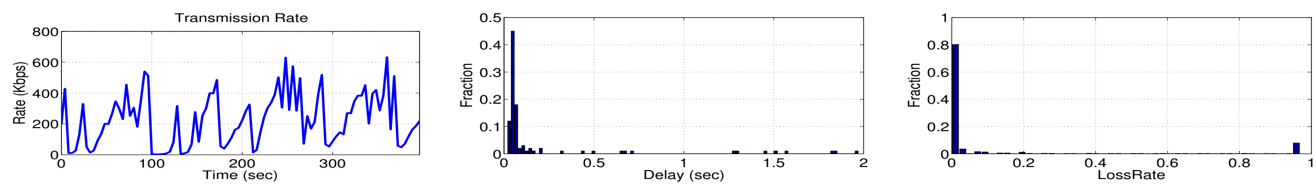
(a) (b) (c)

Fig. 13. (a) Transmission rate, (b) PDF of queuing delay, and (c) PDF of loss rate when using CTCP-like rate control on 3G network.



(a) (b) (c)

Fig. 14. (a) Transmission rate, (b) PDF of queuing delay, and (c) PDF of loss rate when using CTCP-like rate control on WiMAX network.



(a) (b) (c)

Fig. 15. (a) Transmission rate, (b) PDF of queuing delay, and (c) PDF of loss rate when using TFRC on a 3G network.

intensive multimedia applications. such as video conferencing or video on demand. Choosing a very high k_0 and thus a high $\hat{\delta}$ may provide for full link utilization, but will result in poor performance for real-time multimedia applications, such as VoIP or video conferencing. A high $\hat{\delta}$ may also result in congestion induced packet loss which degrades real-time application performance.

Fig. 17 and Fig. 18 show the results when using this algorithm using fixed values of $k_0 = 25000$, $k_2 = 1.6$. From Table I and Table II, we see that improperly selecting the parameters has resulted in too high of a congestion point (too much packet loss), even though the link is fairly well utilized. We note that regardless of what k_0 is used, there may be some network where the choice is too high or too low.

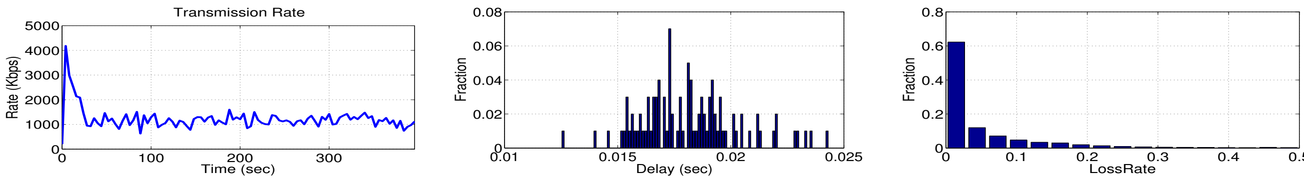
V. Congestion Signals

In Sec. III, we have seen that queuing delay and loss are noisy congestion signals on mobile broadband networks, and in Sec. IV, we have seen that using fixed definitions of congestion result in poor performance.

We naturally ask the question: for 3G and WiMAX networks, what is the best way to distinguish whether we are in the congestion zone or in the congestion-free zone? Should the congestion signal be *queuing delay*, *packet loss*, or some combination of the two? In order to make an informed and in-depth observation, we attempt to classify the queuing delay and loss measurements in Fig. 2 and Fig. 3 into a zone of congestion and a zone of congestion-free.

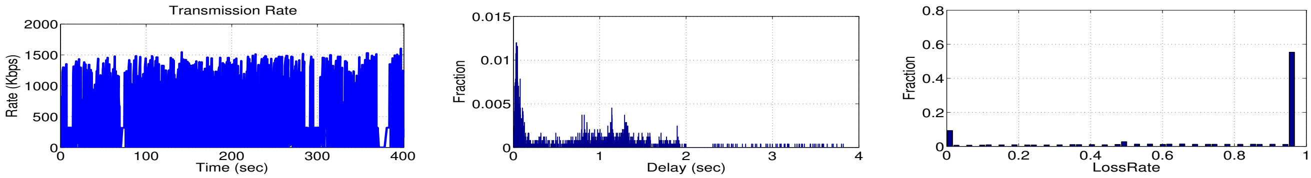
As an example, for the 3G network in the upload direction in Fig. 2, suppose we consider all measurements below 250 kbps to be free of congestion, and measurements from the rate above 640 kbps to be congested. For the WiMAX network in the download direction in Fig. 4, suppose we consider all measurements below 4900 kbps to be free of congestion, and measurements from the rate above 5100 kbps to be congested. The lower rate threshold for each network is chosen by finding the maximal rate whose average delay and loss values are below some threshold. The upper threshold is chosen by finding the minimal rate whose average measurements are above some threshold. We plot the probability density function (PDF) of queuing delay and loss measurement taken from these two sets of rates in Fig. 19.

We observe that in both 3G and WiMAX broadband networks, delay is a very good signal for congestion detection. That is we can pick a delay threshold, δ_T, such that most values from the congestion-free measurement set fall below δ_T and most values from the congested measurement set fall above. For example, for the 3G network, any delay above



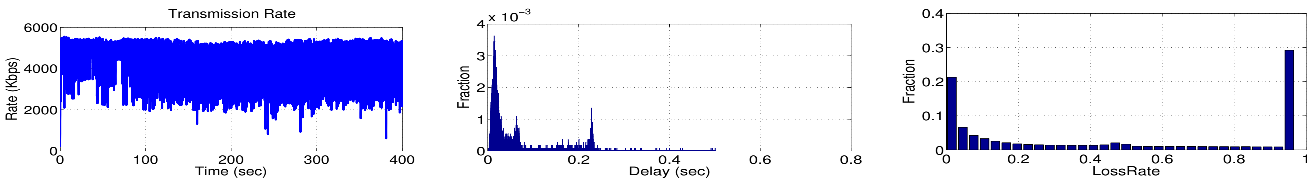
(a) (b) (c)

Fig. 16. (a) Transmission rate, (b) PDF of queuing delay, and (c) PDF of loss rate when using TFRC on a WiMAX network.



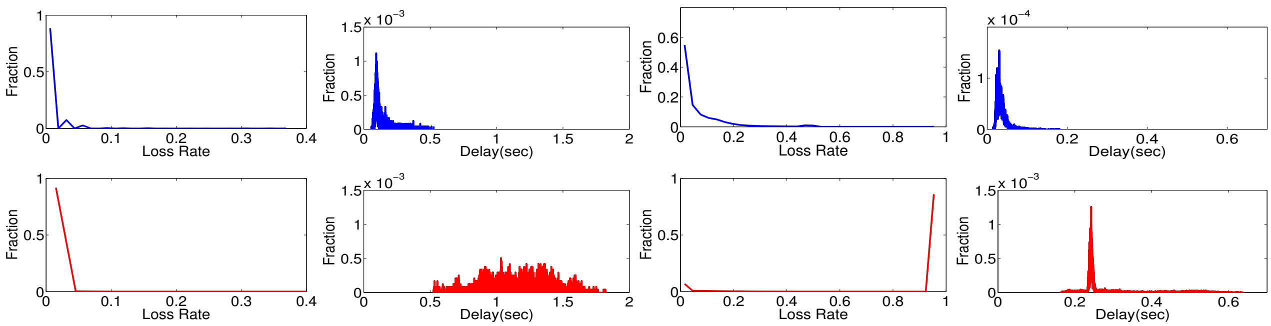
(a) (b) (c)

Fig. 17. (a) Transmission rate, (b) PDF of queuing delay, and (c) PDF of loss rate when using Utility Maximization rate control on 3G network.



(a) (b) (c)

Fig. 18. (a) Transmission rate, (b) PDF of queuing delay, and (c) PDF of loss rate when using Utility Maximization rate control on WiMAX network.



(a) (b) (c) (d)

Fig. 19. The PDF in the congestion-free zone (top figure) and the congestion zone (bottom figure) for (a) loss rate over 3G network, (b) delay over 3G network, (c) loss rate over WiMAX network, and (d) delay over WiMAX network.

300ms can be considered congestion and for the WiMAX network, any delay above 50ms can be considered congestion. Loss is not a good signal for congestion detection. For example on the 3G network, loss characteristics are very similar for both the congestion-free and congested set. In either zone, we may see 2% packet loss. For the WiMAX network, we can choose a loss threshold, ϵ_T such that the two sets are separated. However, even a 5% loss rate is fairly common in the non-congested set and thus the threshold needed is very large (20% loss rate). Thus if we use loss as the congestion signal and desire full link utilization, the operating loss rate will be prohibitively high for real-time applications. Clearly, if we use any loss to be taken as congestion, as TCP NewReno-like rate control does, we will under-utilize the link.

A. Rate Control Using Variable Definition of Congestion

Since delay is an appropriate indicator of congestion, we start from the primal-dual utility maximization based rate control framework presented in Sec. IV-E and present a rate control strategy for multimedia applications. We show that using variable definitions of congestion can result in good link utilization as well as low queuing delay and packet loss ratios. Since we use variable parameters, in order to guarantee fairness, this rate control strategy is suited towards cases where there is a single bottleneck link which is the edge link, as is often the case in mobile broadband networks. If congestion is detected in intermediate links with high capacity and high number of flows, we can fall back to TCP-like strategies.

Consider the rate control in (1). In (1), the operating congestion point, $\hat{\delta}$ is controlled through k_0. A slightly modified version of this is presented in [19], which directly controls the operating congestion point $\hat{\delta}$, and works for single bottleneck link cases. In [19], the rate control used is,

$$\Delta R = \begin{cases} \alpha_\delta & \text{if } \delta \leq \hat{\delta} \\ -\beta_\delta R & \text{if } \delta > \hat{\delta} \end{cases}, \tag{2}$$

where α_δ and β_δ are fixed constants or functions of δ used in the AIMD adjustment. For example, α_δ and β_δ can be linear functions of δ as in

$$\alpha_\delta = \alpha_{\max} + (\alpha_{\min} - \alpha_{\max})\frac{\delta}{\hat{\delta}} \tag{3}$$

$$\beta_\delta = \min\left(\beta_{\max}, \beta_{\min} + (\beta_{\max} - \beta_{\min})\frac{\delta - \hat{\delta}}{\delta_{\max} - \hat{\delta}}\right) \tag{4}$$

Regardless of whether (1) or (2) is used, the operating congestion point is given by $\hat{\delta}$ and is either directly set or indirectly set through the choice of k_0. In order for the rate control strategy to provide the lowest possible congestion point given network characteristics while providing full link utilization, we need to find $\hat{\delta}$ to properly disambiguate the boundary between the *congestion* zone and the *congestion-free* zone.

We propose to adapt or choose $\hat{\delta}$ (either directly or indirectly through k_0) by learning network characteristics for various networks. For example, for clean networks $\hat{\delta}$ can be chosen to be a low value and still result in full network utilization. For noisy networks such as mobile broadband networks, $\hat{\delta}$ needs to be large enough so that the rate control algorithm does not confuse random noise in delay as a congestion signal. At the same time, we do not want to arbitrarily set it to a large value as it hurts real-time multimedia applications. For full link utilization, while maintaining a low congestion level, an appropriate choice is to make $\hat{\delta}$ just slightly larger than δ_T. For example, δ_T can be chosen using the analysis above to minimize the misclassification of elements in the congestion-free set as being congested. The congestion-free set of delay measurements consists of delay values obtained when transmitting at rates which are known to be not congested, for example very low rates.

B. Performance

In this section, we show the performance of the UM based rate control with variable definitions of congestion. We use the rate control algorithm in Eqn. 2 with $\hat{\delta} = 0.4$ for the 3G network and $\hat{\delta} = 0.1$ for the WiMAX network. We compare with the existing schemes presented in Sec. IV. We show the results in Fig. 20 and Fig. 21 and compare with existing schemes in Table I and Table II. Compared with TCP-NewReno-like rate control, TCP-Vegas-like rate control, CTCP-like rate control and TFRC, we see that by learning the proper congestion operating point, we are able to achieve good link utilization at reasonable delay and loss levels, which are close to the 70-80th percentile of the inherent noise level within the network. We see that for the 3G network, we have close to 26% of throughput gain compared to TCP-NewReno-like rate control and similar throughput as CTCP-like rate control but with much lower loss. For the WiMAX network, we see 50% improvement over CTCP-like rate control and standard UM with lower loss and delay and 7x improvement in throughput over TCP-NewReno-like rate control.

In Fig. 22, we also show fairness across two flows. One flow is run from 0-1000 seconds, and a second parallel flow is run from 200-700 seconds. During the period where one flow is running, it obtains approximately the full 5Mbps bandwidth, with a delay of about 20ms. When the second flow enters, they both take about 2.5Mbps each, with the queuing delay remaining close to 20ms. Additionally, unlike standard rate control algorithms with fixed parameters, the delay does not increase when the transmission rate drops. Instead it remains close to the minimum allowed by the inherent noise in the network. This is because our rate control framework tries to achieve a desired operating delay point rather than using fixed parameters. The loss for the entire session is also negligible and stays close to 0.2%. Although convergence currently takes about 20-30 seconds, techniques similar to that used in TCP slow start can be used to speed up convergence.

In Fig. 23, we show the throughput (R) and delay (δ) achieved when using different values of $\hat{\delta}$ using a real implementation of the proposed algorithm over different network configurations. We see that if we choose $\hat{\delta}$ appropriately and larger than δ_T, we can achieve full throughput. However, in order to keep operating delay low, we should choose $\hat{\delta}$ as small as is needed in order to maintain full throughput. Since δ_T is a function of the network, $\hat{\delta}$ should also be chosen for a given network by *learning* and should not be fixed as is done in existing rate control algorithms. By doing this, we can maintain full throughput needed by media applications at low operating congestion levels (low queuing delay) needed for real-time media applications.

VI. Conclusion

In this paper, through the lens of extensive illustrative network traces, we have shown strong evidence that loss and delay are often very noisy signals in mobile broadband networks. When this is the case, existing rate control techniques — such as TCP like rate control schemes, TFRC, and other utility maximization schemes with fixed parameters — either fail to fully utilize the link or fail to function at reasonable queuing delay and loss operating points. This makes media delivery and especially real-time video conferencing very problematic in such networks. The highlight of this paper is our demonstration that the state of congestion can be *learned* relatively easily in mobile broadband networks, and our proposal of a new adaptive algorithm to effectively learn congestion operating points, with significantly improved performance in our real-world experiments.

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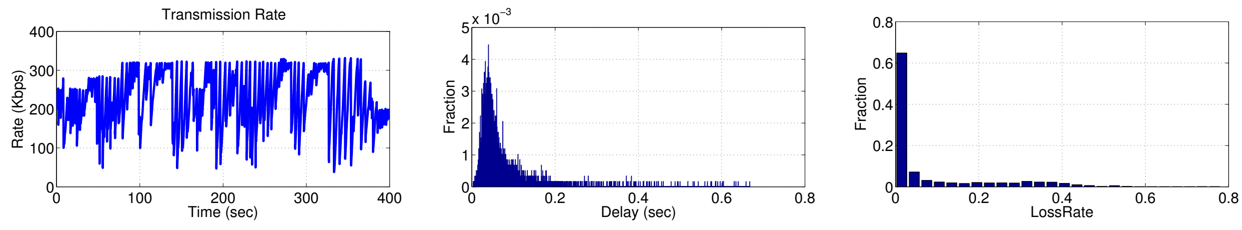
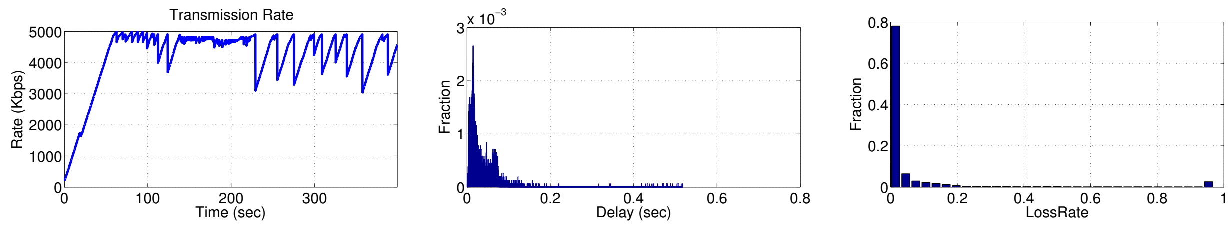


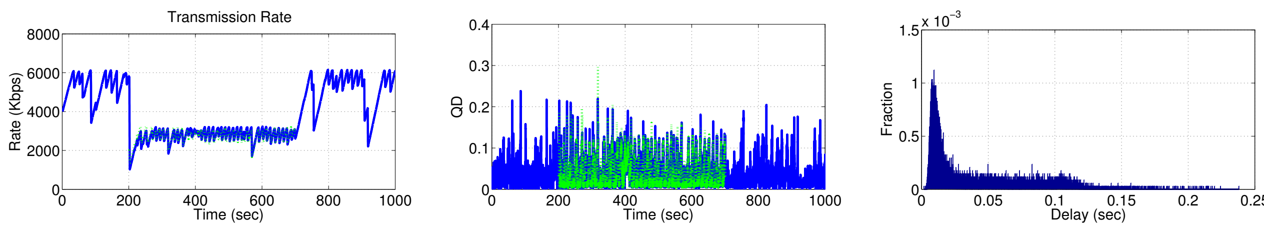
(a) (b) (c)

Fig. 20. (a) Transmission rate, (b) PDF of queuing delay, and (c) PDF of loss rate when using learning based algorithm on 3G network.



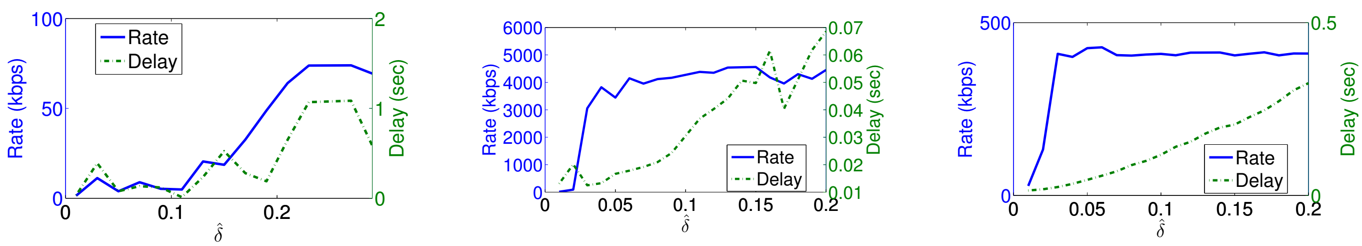
(a) (b) (c)

Fig. 21. (a) Transmission rate, (b) PDF of queuing delay, and (c) PDF of loss rate when using learning based algorithm on WiMAX network.



(a) (b) (c)

Fig. 22. (a) Transmission rate across time, (b) queuing delay across time, and (c) PDF of queuing delay when using a learning based algorithm on WiMAX network with two flows. Two parallel flows are running from 200-700 sec. Overall loss rate is about 0.2%.



(a) (b) (c)

Fig. 23. Rate and operating congestion level (observed queuing delay) as function of $\hat{\delta}$ for (a) 3G download, (b) WiMAX download, and (c) WiMAX upload.

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