## Modeling and Quantifying the Impact of P2P File Sharing Traffic on Traditional Internet Traffic

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### Abstract

Peer-to-Peer (P2P) applications consume most of Internet bandwidth and cause access network congestion. They severely deteriorate the performance of the traditional Internet applications. Previous researches have studied the mechanisms to improve the aggregate throughput of P2P traffic, but there doesn't exist any research to predict and quantify its impact. In this work, we develop a performance modeling system of P2P file sharing traffic and traditional Internet traffic (WEB traffic) to quantify the impact of P2P file sharing traffic on WEB traffic in the congested access network. We answer the following questions from a user's point of view: How many P2P concurrent connections and what proportion between P2P traffic and WEB traffic will guarantee certain WEB performance. The simulation results demonstrate that our model is accurate and efficient.

## 1. Introduction

Peer-to-Peer (P2P) is a large consumer of traffic on Internet Service Providers (ISPs) networks and strongly influences the behavior of other Internet traffic. Now P2P traffic significantly outweighs web traffic and continues to grow. An interrelated study [10] estimates that the aggregate traffic of all P2P applications contribute to about 60-70% of the traffic in the Internet and about 80% of the traffic in the lastmile providers' networks. The fundamental idea of P2P network is to have peers cooperate in an overlay network and operate as both servers and clients, and then the service burden is distributed to participating peers from the burdened servers. P2P technology improves the transmission efficiency and the network utility. On the contrary, it is quite common for peers to run multiple P2P connections concurrently and left them for a long duration. The result is rampaging bandwidth consumption mainly in access network, which is threatening to choke the Internet. Network usage patterns are changing and network capacity provisioned is no longer sufficient, which makes the last-mile network become a congested network bottleneck. P2P traffic causes network congestion and other Internet traffic performance deterioration, which ultimately leads to customers of ISPs dissatisfy and churning.

P2P file sharing applications are more and more popular and attractive, but their aggressiveness leads to network congestion and unfairness. A P2P user gets the file from multiple peers and keeps one or more P2P connections with each peer. As the number of P2P connections increases, P2P applications tend to unfairly steal bandwidth from other Internet applications to deteriorate their performance. However, P2P traffic should not be blocked blindly, which will make ISPs risk losing subscribers. How to effectively guide P2P traffic is urgent. Through quantifying and fine-tuning the flow number of both traffics, P2P traffic will be friendly and controllable.

However, few researches predict and quantify P2P's impact on traditional Internet traffic. We attempt to take a very small step in this direction to quantify the impact of P2P traffic on the traditional Internet traffic.

Before P2P technology appears, Web traffic dominates most of the Internet traffic. So we mainly analyze the impact of P2P traffic on Web traffic. We often meet such a scenario: When someone uses P2P applications, the HTTP performance of this user or other users in the same access network will be badly deteriorated. Based on this scenario, we compare and analyze the impact of P2P traffic on Web traffic in the congested last-mile network. We focus on the P2P file sharing traffic based on TCP protocol, because most of

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P2P file-sharing applications use TCP for accurate and reliable in-order transmission of data. We look on Web traffic as short-lived TCP flow. In some scenarios, UDP-based applications (VoIP or Video) are also a significant portion of Internet traffic. Future work is needed to study the impact of P2P traffic on UDPbased applications. Firstly we propose an integrated model of P2P traffic and Web traffic with multiple connections. Then we estimate the throughput, the loss rate and the round time trip (RTT) of both kinds of flows. Finally we obtain the highest threshold of P2P concurrent connection number and the service proportion between P2P traffic and WEB traffic, in the case of guaranteeing certain WEB performance. These quantitative results will be useful references for ISPs controlling P2P traffic and P2P system designers optimizing the number of P2P connections.

The rest of the paper is organized as follows. In section II, we briefly review some mathematical models of TCP flow. We propose the integrated model of P2P file sharing traffic and Web traffic in section III. In section IV, we validate the integrated model and analyze and quantify the impact of P2P traffic on Web traffic with ns2 simulation. We conclude the paper in the last section.

### 2. Related Work

While early work on P2P systems has mainly focused on system design and traffic measurement, some recent researches have emphasized performance analysis and propose many P2P network models. In [11], a closed queuing system is used to model a general P2P file sharing system and basic insights on the stationary performance are provided. In [12] a fluid model is used to characterize the performance of BitTorrent-like networks in terms of the average number of downloads and the download times. Our work differs from them and analyzes the impact of P2P file sharing on the Internet traditional traffic. Irrelevant to the P2P system mechanism, our integrated model focus on the process of P2P data transfer with multiple TCP connections and the impact of data transfer process on Internet network and tradition traffic.

A lot of researches have been done to develop separate models for bulk transfer TCP and short-lived TCP in order to predict their performance. In [1], an analytical model is firstly developed for the steady state throughput of a bulk transfer TCP flow as a function of loss rate and round trip time. This model captures the behavior of TCP's fast retransmit mechanism as well as the effect of TCP's timeout mechanism. Cardwell [2] extends the steady state model proposed in [1] to capture these startup effects. The extended model characterizes data transfer latency as a function of Web page size, round trip time, and packet loss rate. Some evolution and extension are developed in [4]. Our model differs from them. We study the impact of bulk transfer TCP data transfer on short-lived TCP latency, namely the impact of a bulk of P2P data transfer on the performance of Web transfer.

In [5] the adverse impact of the short-lived TCP flows on the co-existing long-lived TCP flows is studied. In [14] predict parallel TCP throughput as a function of the number of flows, as well as the corresponding impact on cross traffic. Our integrated model has some difference: we study the impact on short-lived TCP performance; we consider multiple P2P connections with different RTT and different senders.

## **3. Integrated Model**

Most of popular P2P file-sharing applications use TCP for accurate and reliable in-order data transfer. In spite of the transient characteristics of the peers such as arrival and leaving of peers, we only analyze the steady state behavior of P2P data transfer in the period of time when the number of P2P connections is changeless. We look on Web traffic as short-lived TCP. Some Web traffics also have characteristics of long-lived TCP flow, which is not analyzed in this paper. We propose a new integrated model to analyze the impact of P2P traffic on Web traffic in the congested last-mile network. Our integrated model is the integration and extension of [1] [2] [4] and has exactly the same assumptions about the endpoints and network.

First, we assume that the sender uses a congestion control algorithm from the TCP Reno family; We assume that the receiver uses delayed acknowledgment (ACK), whereby it sends an ACK for every b=2 data segments; We don't consider the receiver window size limitation, because the experiment is in congested access network and the congestion window size will be less than the receive-window size. We assume that ACK loss can be neglected because ACKs size is relatively small. We just consider the bottleneck router with RED. RED has the potential to overcome some of the problems discovered in Drop-Tail such as synchronization of TCP flows, correlation of the drop events within a TCP flow and avoid the bias against burst traffic in [9]. So we assume the loss rate of both kinds of traffic is equal in the RED router.

#### 3.1. P2P traffic Model

We focus on the process of data transfer, then. Padhye et al. [1] developed a complete model for the steady state throughput of a bulk transfer TCP flow as a function of loss rate and round trip time. This model captures the behavior of TCP's fast retransmit mechanism as well as the effect of TCP's timeout mechanism. The model can be stated as:

$$B(p) \approx \frac{MSS}{RTT\sqrt{\frac{2bp}{3} + T_o \min\left(1, 3\sqrt{\frac{3bp}{8}}\right)p(1+32p^2)}} = \frac{MSS}{RTT \times f(p)} (1)$$
  
Where  $f(p) = \sqrt{\frac{2bp}{3} + T_o \min\left(1, 3\sqrt{\frac{3bp}{8}}\right)p(1+32p^2)} (2)$ 

Where *p* is the packet loss rate, *RTT* is the round trip time of TCP.  $T_0$  is the initial value of the timeout and b is the number of packets that are acknowledged by a received ACK.

In a P2P network with n multiple TCP flows, the aggregate bandwidth of all n TCP connections in the access network bottleneck can be stated as following:

$$B(p)_{p_{2}p} \le \sum_{i=1}^{n} \frac{MSS_{i}}{RTT_{i} \times f(p_{i})}$$
(3)

We assume that each MSS value is identical and constant across all simultaneous TCP connections between hosts. In P2P network, a P2P host establishes TCP connections with different peers, so the paths of P2P connections are distributed and the RTTs of P2P connections are heterogeneous. We let the heterogeneous RTT of the *ith* TCP flows be  $RTT_i$ . In [6], the packet loss distribution among the parallel TCP is discussed in detail. To avoid the unfair distribution of packet loss in congested router, some queuing schemes such as Random Early Detection (RED) in [8] are proposed and deployed. We assume that packet loss pof P2P connections and the average waiting time  $T_{wait}$  are equally in the congested access network with RED mechanism.

Thus, equation (3) can be modified as following:

$$B(p)_{P2P} \leq \frac{MSS}{f(p)} \left( \frac{1}{RTT_1} + \frac{1}{RTT_2} + \dots + \frac{1}{RTT_n} \right)$$
(4)

We consider the scenario of congested network, then

$$B(p)_{p_{2}p} = \frac{MSS}{f(p)} \left( \frac{1}{RTT_1} + \frac{1}{RTT_2} + \dots + \frac{1}{RTT_n} \right)$$

$$= \frac{MSS}{f(p)} \sum_{i=1}^n \frac{1}{RTT_i}$$
(5)

### 3.2. Web traffic Model

We look on Web traffic as short-lived TCP flow. Some Web traffics with the characteristics of longlived TCP flows are not considered in this paper. In [2] [3] [4], the short-lived TCP model is an extensive stochastic model which can more accurately predict the throughput and latency of short-lived TCP connections. This model is composed of four parts: the start of the  $E[d_{ss}]$  connection (three-way-handshake), the initial slow-start, the first loss part, and the subsequent losses.

Let  $E[d_{ss}]$ ,  $E[W^{ss}]$  and H be the throughput of three phases respectively, then

Ε

$$E[d_{ss}] = \frac{(1 - (1 - p)^d)(1 - p)}{p} + 1$$
(6)

$$\left[W^{ss}\right] = W^{ss} \tag{7}$$

$$H = \frac{\frac{E[W^{TD}]g^2}{2} - 2 + \frac{1}{Q^{TD}[E[W^{TD}]]} \left(\frac{1-p}{p} + (E[W^{TD}] - 1](1-p)\right) + 1}{\left(\log\left(\frac{E[W^{TD}]}{2C_1}\right) + \frac{1}{Q^{TD}[E[W^{TD}]]} \left(\frac{bE[W^{TD}]}{2} + b + 1\right)\right) RTT_{Web}} + \frac{f(p)T_o}{1-p}$$
(8)

Where  $d_{ss}$  is the number of data segments before losing a segment. *d* is the WEB page size.  $E[W^{ss}]$  is the window we would expect TCP to achieve at the end of the slow start.

 $E[W^{TD}]$  is the expected congestion window size in the congestion avoidance phase:

$$E[W^{TD}] = -\frac{2(b-2p)}{3} + \sqrt{\frac{4(bp+2(1-p^2))}{3bp}} + \left(\frac{2b-4p}{3b}\right)^2 (9)$$
  
Where  $f(p) = 1 + p + 2p^2 + 4p^3 + 8p^4 + 15p^5 + 32p^6$ ,  
 $g = \frac{1+\sqrt{5}}{2}$ ,  $Q^{TD}$  is the probability that a loss detection  
is a time out, given as  $Q^{TD} \approx \min\left(1, \frac{3\sqrt{3}}{E[W^{TD}]}\right)$ .

We decompose the data transfer latency,  $T_{latency}$ , for *d* data segments into five aspects: the connection established phase  $E[T_{nohs}]$ , the initial slow start phase  $E[T_{ss}]$ , the resulting packet loss  $T_{loss}$ , the transfer of remaining data  $T_{rest}$ , and the added delay from delayed acknowledgments  $T_{delay}$ , then

$$E[T_{twhs}] = RTT_{Web} + T_s \left(\frac{1-p}{1-2p} - 1\right)$$
(10)

$$E[T_{ss}] = RTT_{Web} \times \left[\log_g\left(\frac{E[d_{ss}] + 2}{C_1}\right)\right] - 2$$
(11)

$$T_{loss} = (1 - (1 - p)^d) (Q^{ss} E[Z^{TO}] + (1 - Q^{ss}) E[n_t])$$
(12)

$$T_{rest} = \frac{d - E[d_{ss}]}{H} \tag{13}$$

Where  $T_s$  is the duration of SYN time-out, its initial value is 3s [7]. In (11),  $C_1 = \frac{5 + \sqrt{5}}{10}$ . In (12),

$$Q^{ss} = \begin{cases} 1, & \text{if } M = 2\\ \sum_{k=0}^{1} A(W^{ss}, k) + \sum_{k=2}^{W^{ss}-1} A(W^{ss}, k) h\left(\left\lfloor \frac{k}{2} \rfloor + k\right), \text{ otherwise} \end{cases}$$

 $A(w,k) = \frac{(1-p)^k p}{1-(1-p)^w}$ ,  $W^{ss} = \frac{E[d_{ss}]+2}{g^2}$ , the expected

cost of an RTO is  $E[Z^{TO}] = \frac{T_o f(p)}{1-p}$ . When the

congestion window size  $W^{ss}$  is bigger than three, the expected time can be calculated as:

$$E[n_{t}] = RTT_{short} \times \frac{1 - (1 - p)^{W^{ss} - 3} - (1 - p)^{W^{ss}}}{1 - (1 - p)^{W^{ss}}}$$

 $T_{delay}$  is the expected delay between the reception of a single segment and the delayed ACK for that segment 100ms for BSD-derived stacks, and 150ms for Windows. Grouping above formulas together, we now have the total expected latency of short-lived TCP:

$$T_{Latency} = T_{twhs} + T_{ss} + T_{loss} + T_{rest} + T_{delay} - \frac{RTT_{Web}}{2} (14)$$

The average throughput of Web traffic is estimated:

$$B(p)_{Web} = \frac{d}{T_{latency}}$$
(15)

# **3.3.** The Integrated Model of P2P traffic and Web traffic

We combine the P2P traffic model with Web traffic model in the bottleneck router with RED algorithm. We assume that the number of P2P flows (n) and the number of Web flows (m) don't change dramatically over a period of time.

Let  $B(p, RTT_j)_{w_{cb}}$  be the *j*th short-lived TCP flow respectively. Then the aggregate throughput of *n* P2P flows referring to (5) and *m* Web flows on the congested network bottleneck router equals to the network bottleneck capacity.

$$u \approx \frac{MSS}{f(p)} \sum_{i=1}^{n} \frac{1}{RTT_i} + \sum_{j=1}^{m} B(p, RTT_j)_{Web}$$
(16)

The average round trip time of a TCP flow equals to the sum of the average waiting time  $T_{wait}$  in the queue of the single bottleneck router, and the propagation time  $\tau$ . We assume the forwarding policy of RED router to the P2P flows is the same as Web flows. So the average waiting time of P2P flows is also the same as Web flows, equaling to  $T_{wait}$ . We let the propagation time of the *i*th P2P flow path be  $(\tau_i)_{P2P}$ , and the propagation time of the *j*th Web flow path be  $(\tau_i)_{Web}$ ,

$$\left(RTT_{i}\right)_{P2P} = \left(\tau_{i}\right)_{P2P} + T_{wait}$$

$$(17)$$

$$\left(RTT_{j}\right)_{Web} = \left(\tau_{j}\right)_{Web} + T_{wait} \tag{18}$$

Using the average queue length, the RED algorithm calculates a packet marking probability at every arrival of an incoming packet as:

$$p = \begin{cases} 0, & 0 \le \bar{q} \le q_{\min} \\ \frac{\bar{q} - q_{\min}}{q_{\max} - q_{\min}} p_{\max}, & q_{\min} \le \bar{q} \le q_{\max} \end{cases}$$
(19)  
1,  $q_{\max} \le \bar{q} \le B$ 

Where B is the buffer size,  $q_{\min}$ ,  $q_{\max}$  and  $p_{\max}$  is the control parameters of the RED. The  $\bar{q}(p)$ , average

queue length, must be less than or equal to the router's buffer size Buf. Then we have the following expression for the average queue size as a function of the drop probability p:

$$\bar{q}(p) = \max\left(Buf, \frac{p(q_{\max} - q_{\min})}{p_{\max}} + q_{\min}\right)$$
(20)

Using (16), the average waiting time in the bottleneck router can be calculated as:

$$T_{wait} = \frac{q(p)}{u} \tag{21}$$

Then combining with the formula (5) (15) (16) (17) (18) (20), we can get the integrated formula (22) for the bottleneck capacity as a function of loss rate, P2P traffic connection number *n* and Web traffic connection number *m*. With certain values of *n* and *m*, We can calculate  $B(p)_{P2P}$ ,  $B(p)_{Web}$ ,  $T_{latency}$ ,  $RTT_{P2P}$  and  $RTT_{Web}$  using the calculated value *p* from (22).

$$u \approx \frac{MSS}{f(p)} \sum_{i=1}^{n} \frac{1}{(\tau_{i})_{p_{2}p} + \frac{\bar{q}(p)}{u}} + \sum_{j=1}^{m} B\left(p_{j}(\tau_{j})_{w_{eb}} + \frac{\bar{q}(p)}{u}\right)_{w_{eb}} (22)$$

In conclusion, through regulating the number of P2P flows *n*, the number of Web flows *m*, and the parameters of  $\tau_{P2P}$ ,  $\tau_{Web}$ ,  $q_{\min}$ ,  $q_{\max}$  and  $p_{\max}$ , we can obtain the performance of P2P traffic and Web traffic in a single bottleneck router to analyze and understand the impact of P2P traffic on Web traffic.

### 4. Analysis and Simulation

To validate our proposed integrated model and analyze the impact of P2P traffic on Web traffic, we perform mathematic calculation with MATLAB and simulation experiments with NS2. In the simulation, we just consider the P2P file transfer process and performance, so we look on the P2P traffic as bidirectional FTP application with many senders and a receiver.



We run the simulation experiments for the same network topology given by Fig. 1. We have performed three simulation sets when the capacity *u* of access link *L* are *1Mbps*, *10Mbps*, *100Mbps* respectively. We assume only a single bottleneck point *L* in the network. Web traffic is from the Web server outside of the access network to client in the access network. P2P traffic is between the hosts in the access network and the peers outside of the access network. For each scenario, we perform several experiments with different *n* and *m*. The propagation time,  $\tau_{long}$  and  $\tau_{short}$ , are set as a random value in interval (*10ms*, *100ms*). The average packet size is set as 500bytes. The buffer at link *l* has a size of BDP to avoid packet drops due to buffer overflow.

P2P traffic simulated by some "infinite" FTP applications exists in the whole simulation. At a certain moment, m Web applications are triggered, then we record the beginning time and the end time of Web traffic and calculate the Web latency to compare them with the results of our integrated model. A large number of simulations have been done to gain an average value of Web latency with a certain proportion of n:m.

In the first group of simulations, we simulate the scenario where n P2P connections and *one* Web connection coexist to calculate the Web latency and to show the max threshold of P2P connections. Then in second group, we quantify the impact of P2P traffic on Web traffic in different network scenarios with different aggregated connection number of P2P traffic on Web traffic.

#### 4.1. The Max Number of P2P Connections

The simulations are in the scenario that n P2P connections and *one* Web connection coexist to calculate the Web latency to find the max number of P2P connections while assuring a certain performance

of Web service (for example, webpage size = 20kbytes, latency = 5s). Then we compare the simulation results with our integrated model results to validate the integrated model.



**Fig. 2.** The latency when n P2P and 1 WEB coexist

Fig. 2 shows the relationship between the latency and the number of P2P connections in different network scenarios. The solid lines are the values calculated by the integrated model. The dots are the values from the simulations. While the number of P2P traffic increases, Web performance deteriorates. For example, a max threshold of P2P connections can be found as 22 to guarantee the web latency as 5s when the network bottleneck bandwidth is 1Mbps. From the figures, it is clear that the values predicted by our model match the simulation values well.

 
 Table I the relationship of Web latency and the max threshold of P2P connections MAXn

Web performance level	excellent	good	bad	unusable
Web latency value (s)	5	30	60	>60
The <i>MAXn</i> when <i>u</i> =1Mbps	22	446	975	>975
The MAXn when u=10Mbps	530	2620	3700	>3700
The MAXn when u=100Mbps	4300	1269 0	1416 5	>14165

If we classify the web performance to four levels (excellent, good, bad, and unusable) as Table I, the max threshold of P2P connections can be found when a certain Web performance is guaranteed in a certain network scenario. These thresholds will change according to some parameters variety in the experiment network. The quantified results will be helpful for ISPs and P2P program designers to control P2P traffic and optimize the number of P2P connections.

# 4.2. The latency and the connection number proportion of P2P and Web

Fig. 3 shows the relationship of the latency and the proportion of P2P traffic and Web traffic, while the aggregated connection number is fixed.



(c) *u*=100Mbps **Fig. 3.** When aggregated connection number=50, the relationship of web latency and P2P connections number

As can be observed, to guarantee the latency<5s, the connection number proportion should under 17:33 when u=1Mbps. It is also clear that the values predicted by our model match the simulation values.

### 5. Conclusion

This paper presented a study on how P2P applications impact on traditional Internet applications in the congested access router with RED algorithm. We develop an extensive integrated model for predicting the performance of Web application in terms of the latency when P2P traffic and Web traffic coexist. This model quantitatively analyzes the impact of P2P traffic on WEB traffic, estimates the throughput, the loss rate and the round time trip (RTT) of both kinds of traffic, as well as the WEB latency, even if they coexist in different proportion. With the model we can obtain the highest threshold of P2P concurrent connection number and the service proportion between P2P traffic and WEB traffic, when guarantying certain WEB performance. Finally we validated the integrated model with simulation experiments. The results show that our model is accurate and efficient. We believe that the quantified

results will be helpful for ISPs and P2P program designers to control P2P traffic, optimize the number of P2P connections and make a better convergence of P2P traffic and Internet traditional traffic.

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