A Performance Study of a Large-scale Data Collection Problem*

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Abstract

In this paper, we consider the problem of moving a large amount of data from several source hosts to a destination host over a wide-area network, i.e., a large-scale data collection problem. This problem is important since improvements in data collection times in many applications such as wide-area upload applications, high-performance computing applications and data mining applications are crucial to performance of those applications.

Existing approaches to the large-scale research are transferring data either directly, i.e., direct methods, or using "best"-path type of application-level re-routing techniques, which we refer as non-coordinated methods. However, we believe that in the case of large-scale data collection applications, it is important to *coordinate* data transfers from multiple sources. More specifically, our coordinated method would take into consideration the transfer demands of all source hosts and then schedule all data transfers in parallel by using all possible existing paths between the source hosts and the destination host.

We present a performance and robustness study of different data collection methods. Our results showed that coordinated methods can perform significantly better than non-coordinated and direct methods under various degrees and types of network congestion. Moreover, we also showed that coordinated methods are more robust than non-coordinated methods under inaccuracies in network condition information. Therefore, we believe that coordinated methods are a promising approach to large-scale data collection problems.

1 Introduction

The large-scale data collection problem we considered is the problem of moving a large amount of data from a set of source hosts to a destination host over a wide-area network. Such problems correspond to a set of important applications. These applications include online submission of income tax forms, submission of papers to conferences, Internet-based storage, and many more. We refer to such applications as wide-area uploads. Upload applications are likely to become significant contributors to Internet traffic in the near future, as large-scale data transfer applications take advantage of the proliferation of the Internet in society and industry. For instance, consider the online submission of income tax forms, U.S. Congress has mandated that 80% of tax returns to be filed electronically by 2007. With (on the order) 100 million individual tax returns filed in U.S. yearly, where each return in on the order of 100KB [10], scalability and performance issues are a major concern.

In addition to wide-area upload applications, one could consider other applications where improvements in data collection times is an important problem. One example is high-performance computing applications where large amounts of data need to be transferred from one or more data repositories to one or more destinations, where computation on

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that data is performed. Another example is data mining applications where large amounts of data may need to be transferred to a particular server for analysis purposes.

Recently, we proposed a scalable and secure application-level architecture for wide-area upload applications, termed Bistro [4][1]. Given a large number of clients that need to upload their data by a given deadline to a given destination server, Bistro breaks the upload process into three steps: (1) a timestamp step which ensures that the data is submitted on-time *without* having to actually transfer the data, (2) a data transfer step, where clients *push* data to intermediate hosts (termed bistros) participating in the Bistro architecture, which ensures fast response time for the clients, and (3) a data collection step, where a destination server (termed destination bistro) *pulls* data from bistros, i.e., the destination server determines how and when the data is transferred from the bistros. During Step (2) receipts corresponding to clients' transfer are sent by the (intermediate) bistros to the destination bistro; hence the destination bistro knows where to find all the data which needs to be collected during Step (3). We consider Step (3) as a large-scale data collection problem, and the performance of this step is the focus of our paper.

One simple approach to solving the data collection problem is to transfer the data *directly* from source hosts to the destination host based on a transfer schedule computed by the destination host; Hence the data would follow an IP-prescribed route. However, this might result in long transfer times, for example, due to network congestion. Given the current state of IP routing, congestion conditions alone may not necessarily cause a change of routes between a pair of hosts, even if alternate routes exist.

Another approach is to transfer the data using *application-level re-routing* techniques. Recent works such as Detour and RON [2, 14] consider frameworks for application-level re-routing around congestion and failures. Most such re-routing techniques use "best"-path type rerouting, where data from a source host is transferred over the "best" application level path to the destination host. This path is determined independently for each source host (refer to Section 3 for more details).

However, we believe that in the case of large-scale data collection applications, it is important to *coordinate* data transfer from multiple sources. More specifically, the difference from an application level re-routing approach, which we refer to as a non-coordinated method, is that a coordinated method would (i) take into consideration the transfer demands of all source hosts, and (ii) potentially use all possible paths existing between the source hosts and the destination host, all at the application layer. We have been developing such a coordinated approach in the context of our Bistro system (refer to Appendix B for more details). We also note that in the case of Bistro, there is no need for a distributed algorithm for a coordinated approach to the large-scale data collection problem since in Bistro we employ a server pull approach (as stated above), with all information needed to solve the data collection problem available at the destination server.

The contribution and focus of this paper are as follows. We present a study which compares the performance and robustness of these three potential approaches to large-scale data transfer, namely direct, non-coordinated, and coordinated approaches. (The specific performance metrics used in this comparison are defined in Section 4.) We show that coordinated methods can perform better than non-coordinated and direct methods under various degrees and types of network congestion. These improvements are achieved without significant effects on other network traffic throughput. In addition, we show that coordinated methods are more robust than non-coordinated methods under inaccuracies in network condition information.

The remainder of this paper is organized as follows. In Section 2 we briefly survey related work. Section 3 gives a brief overview of the direct, the non-coordinated, and the coordinated methods for the large-scale data collection problem. In Section 4 we give a quantitative evaluation and comparison of these methods. Section 5 gives our concluding remarks.

[1]A more detailed description of Bistro is included in Appendix A.

2 Related Work

As stated above, in this paper we focus on a performance and robustness study of different approaches to large-scale data transfers over wide-area networks, in the context of upload applications. In [4] an application-level *framework* for large-scale upload applications, termed Bistro, is proposed. To the best of our knowledge no other large-scale *upload* architecture exists to date. Hence, we do this work in the context of Bistro, although as noted above, other types of applications can benefit as well. Specifically, we focus on the performance of the *data collection step* as described in Section 1.

We note that some work exists on efficient design of multipoint-to-point aggregation mechanisms at the IP layer, e.g., [3, 5]. However, the solutions suggested there, in the context of IP routers and active networks, are focused on reduction of overheads due to small packets, such as *ACK*s, and aggregation of multiple small packets with the same destination into one large packet. It also requires the use of the active network framework which is not currently widely deployed over the public Internet.

We now briefly survey representative works on application level re-routing. Re-routing at the application level has been used to provide better end-to-end performance or efficient fault detection and recovery for wide-area applications. For instance, in [15] the authors perform a measurement-based study of comparing end-to-end quality of default routing vs alternate path routing (using metrics such as round-trip time, loss rate, and bandwidth). Their results show that in 30% to 80% of the cases considered, there is an alternate path with significantly superior quality. This work provides evidence for existence of alternate paths which can outperform default Internet paths.

Other frameworks or architectures which consider re-routing issues include Detour [14] and RON [2]. The Detour framework [14] is an informed transport protocol. It uses sharing of congestion information between hosts to provide a better “detour path” (via another node) for applications in order to improve the performance of each flow and the overall efficiency of the network. Detour routers are interconnected by using tunnels (i.e., a virtual point-to-point link); hence Detour is an in-kernel IP-in-IP packet encapsulation and routing architecture designed to support alternate-path routing. This work also provides evidence of potential long-term benefits of “detouring” packets via another node by comparing the long-term average properties of detoured paths against Internet-chosen paths.

The Resilient Overlay Network (RON) [2] is an architecture which allows distributed Internet applications to detect failure of paths (and periods of degraded performance) and recover fairly quickly by routing data through other (than source and destination) hosts. It also provides a framework for the implementation of expressive routing policies.

We note that the above mentioned re-routing works focus on architectures, protocols, and mechanisms for accomplishing application-level re-routing through the use of overlay networks. These works also provide evidence that such approaches can result in significant performance benefits.

In this work, we consider a similar environment (i.e., application-level techniques in an IP-type wide-area network). Moreover, we consider an alternate approach, namely that of a *coordinated* data transfers. That is, in above mentioned works all data transfers are treated independently, and hence each takes the “best” application-level route available. In contrast, we also consider coordination of multiple data transfers destined for the same host. We note that this coordination was initially motivated by large-scale data transfer problems which exist in the upload applications mentioned in Section 1. Furthermore, an important focus in this work and its main contribution is the performance and robustness study and the subsequent comparison of different approaches to large-scale transfers, namely direct, non-coordinated, and coordinated approaches.

3 Overview of Data Collection Approaches

In this section, we give a brief overview of the data collection methods evaluated in this paper.

Direct Method
A simple approach to the large-scale data transfer problem is to devise a schedule for transferring the data directly from the source hosts to the destination host, where the data would follow the IP-prescribed route. For instance this

includes:

1. transferring the data from all source hosts to the destination host in parallel, or
2. transferring the data from the source hosts to the destination host sequentially in some order, or
3. transferring the data in parallel from a constant number of source hosts at a time and possibly during a predetermined time slot,

as well as other variants.

Our extensive evaluation of a variety of direct methods showed that the first method given above performs as well as any of the other direct methods [6]. Hence, we use that method as a representative *direct method* in the remainder of the paper.

Non-Coordinated Method
We consider a non-coordinated approach in which each source host chooses a single application-level ("nearly") best path (in term of available bandwidth) to transfer its data to the destination host. Specifically, this non-coordinated method is similar[2] to the current approaches used in RON [2] and Detour [14]. As suggested in [2, 14], in this non-coordinated approach, we only consider re-routing traffic through at most one intermediate host on the way to the destination host. Moreover, for each source host, we first choose a path, p, with one intermediate host, and consider p as the current "best" path. We then consider another potential path, q, and choose q instead of p, if q's available bandwidth is more than 105% of p's available bandwidth, as in [2]. We continue in this manner, until all paths have been considered.

Coordinated Method
To construct a coordinated data transfer schedule, we use a graph-theoretic approach, whose details are given in [6][3]. The basic idea behind this approach is to devise a coordinated data transfer schedule while optimizing a given performance metric, such as makespan (i.e., the total time to transfer the data). This is done by transferring data in parallel over multiple paths with the goal of shortening the makespan metric and scheduling each transfer at some appropriate time, so as to avoid competition for bandwidth among source hosts. For example, we delay some data transfers at some source hosts until an appropriate time, as computed by our graph theoretic approach. Moreover, this coordinated data transfer method is an application level approach and hence does not require any additional information beyond what the above mentioned non-coordinated approaches use.

Figure 1 gives a high-level example to illustrates the intuition behind the differences among these approaches. Assume that both source hosts S1 and S2 have 1 unit of data to upload to the destination host D. Normally, this would go through network N1. Let's further assume that there is a bottleneck link somewhere between N1 and D and this link has unit capacity. Therefore, if D pulls data simultaneously from both S1 and S2. It will take 2 units of time to complete the transfer.

Another host, denoted by K, can transfer data to D at twice the unit capacity. However, it's connection to hosts S1 and S2 is limited at unit capacity. If both S1 and S2 choose to use K to reroute the data to D, it will take 2 units of time to transfer data to K and another unit of time to transfer data from K to D. The total transfer time is 3 in this case. Chances are, uncoordinated approaches will not choose this solution.

However, if we *coordinate* the data transfers in the following fashion, we can cut the total data transfer time to 1.5. During the first unit of time, S1 transfers its data directly to D and, *simultaneously*, S2 transfers its data to K. At the end of this time period, there is 1 unit of data left on K. It would take another 0.5 unit of time to complete the data transfer. As far as the makespan metrics is concerned, the data transfer from S2 to K is *free* because it occurs *in parallel* with the data transfer from S1 to D. Also, it does not *compete* with the other data transfer. This parallelism

[2]We are not able to use the exact re-routing algorithm in [2, 14] as the *details* of these algorithms are not stated there.
[3]A brief overview of this approach is included in Appendix B.

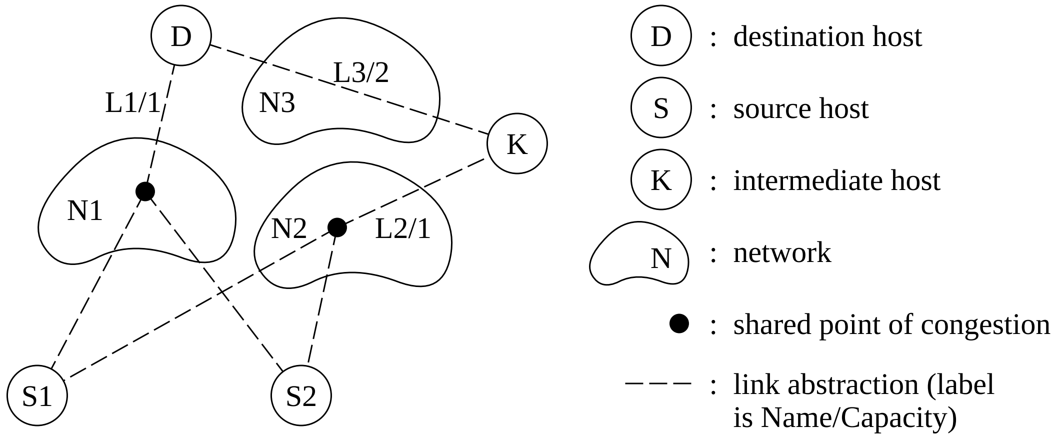


Figure 1: A high-level example.

cannot be exploited explicitly with other approaches. We believe that in a large network, there is a lot of parallelism that can be exploited using our approach.

4 Performance Evaluation

In this section we evaluate the performance of direct, non-coordinated, and coordinated approaches to illustrate the benefits of using coordinated approaches. This evaluation is done through simulation; all results are given with at least $95\% \pm 5\%$ confidence.

Experimental Setup
We use ns2 [8] for all simulation results reported below. Our network topology is generated by the Georgia Tech Internetwork Topology Generator (GT-ITM) [9]; this topology consists of 152 nodes and two transit domains. Each transit domain has, on the average, 4 transit nodes with there being an edge between each pair of nodes with a probability of 0.6. Each node in a transit domain has, on the average, 3 stub domains connected to it. Each stub domain, on the average, has 6 nodes with there being an edge between each pair of nodes with a probability of 0.2. Figure 2 shows the transit domains of our topology. The capacity of transit-stub links and stub-stub links is 2.5Mbps,

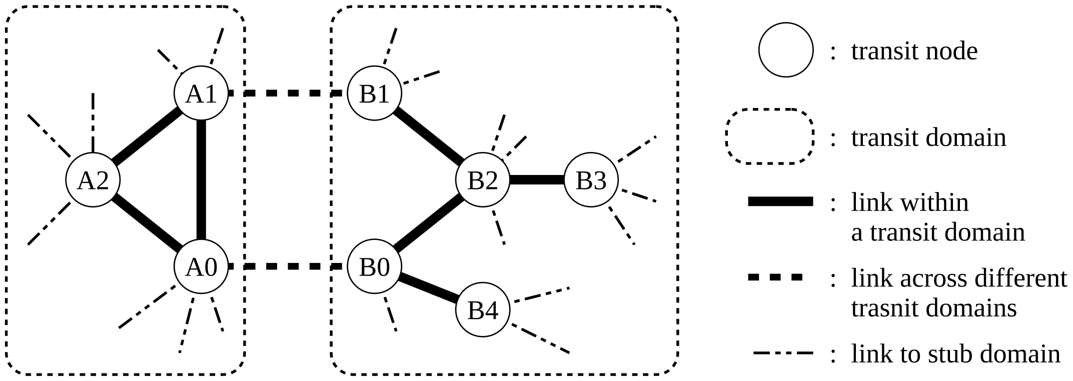

A1
A2
A0
B1
B2
B3
B0
B4
: transit node
: transit domain
: link within a transit domain
: link across different trasnit domains
: link to stub domain

Figure 2: The simulation topology.

while the capacity of inter-transit-domain links (peering points) and intra-transit-domain links are 5Mbps and 10Mbps, respectively. We assign lower capacity to the so-called peering points as it is believed that peering points tend to be more congested [11]. Each source host holds a total amount of data which is uniformly distributed between 25MB to 75MB with an additional constraint that the total amount of data in all 7 source hosts is 350MB. The destination host and 2 of the source hosts are connected to transit domain A, and the remaining source hosts are connected to transit domain B.

We setup 0 to 60 background traffic flows from nodes attached to transit domain B to nodes attached to transit domain A. We investigate how the methods behave under different levels of congestion, which we create through the use of background traffic. Specifically, we control the ratio of the number of background flows in peering point (B0,A0) to the number of background flows in peering point (B1,A1), which in our experiment are 1:1 (symmetric), 2:1, and 1:3 (asymmetric). The background traffic is similar to that in [13]. Each background flow is an infinite FTP with a probability of 0.75. Otherwise, it is a on-off CBR UDP flow. The average on time and off time is chosen uniformly between 0.2 and 3 seconds. The average rate (including the off periods) is chosen so that the expected total volume of UDP traffic through a peering point takes up 0%, 5%, or 20% of the capacity of that point. To measure the available bandwidth of a path, we send (several times) a 5MB file between all pairs of source and destination hosts before transferring any real data. Note that the method of estimating available bandwidth is a separate issue from the coordinated and non-coordinated approaches. We can substitute sending a large file with any available bandwidth estimation approaches, e.g., as in [7, 1, 12][4].

Performance Study
The performance metrics used in the remainder of this section are (a) makespan, i.e., the time needed to complete transferring all data from all source hosts to the destination host, (b) maximum storage requirements averaged over source hosts (not including the destination host since it must collect all the data), and (c) mean throughput of background traffic during the data collection process, i.e., we also consider the effect of upload traffic on other network traffic. We believe that these metrics reflect the quality-of-service characteristics that would be of interest to large-scale data collection applications[5].

Figures 3(c) and 3(d) show the effect of symmetric and asymmetric traffic, under the makespan metric where 5% of the capacity of the peering points is contributed by UDP. The coordinated method performs better than other methods, especially for a large number of background flows, because it coordinates all transfers, and utilizes available bandwidth better. The improvement in the symmetric case is more than 100% over other methods. In the asymmetric case, the makespan of the coordinated method results in 156% and 64% better performance than that of the direct method and the non-coordinated method, respectively. Note that under the symmetric condition, the direct method performs as well as the non-coordinated method. This occurs because in the direct method, the IP-prescribed route of half of the data passes through one peering point, while the IP-prescribed route of another half of the data passes through another peering point. This happens to be a good schedule in this case. Under the asymmetric condition, the non-coordinated method is better than the direct method since it re-routes data around the more congested path[6].

We also ran experiments with ratios of loading on peering points of 1:3, 2:1, and 3:1 (asymmetric). Qualitatively the results are similar, i.e., the coordinated method performs significantly better than the non-coordinated method over various types of network congestion.

[4]We use this method to emulate the use of data collected during previous transfers in estimating bandwidth for future transfers. As already noted, other methods for bandwidth estimation exist [7, 1, 12]. However, most such methods are fairly inaccurate. Hence, we perform a sensitivity study to bandwidth estimation (see the Robustness Study below).

[5]We note that we performed a suite of experiments corresponding to this metric as well, and the results exhibit the same trends as the makespan metric. We do not include these due to lack of space.

[6]Lastly, we note that when the available bandwidth of one path dominates, by a great amount, the available bandwidth of other paths, the performances of the coordinated method and the non-coordinated method should converge since both of them will use that extremely good path to transfer their data.

Effect of UDP traffic
We investigate the effect of UDP traffic on the makespan metric. In Figures 3(a) and (b), 3(c) and (d), and 3(e) and (f) show the makespan metric under 0%, 5%, and 20% of UDP traffic, respectively. Figures 3(a), 3(c), and 3(e) show the symmetric case (1:1) and Figures 3(b), 3(d), and 3(f) show the asymmetric case (2:1). As we can see, Figures 3(a), 3(c), and 3(e) are very similar, so as Figures 3(b), 3(d), and 3(f). Qualitatively, the observations we made above, in comparing direct, non-coordinated, and coordinated methods, still hold under different volumes of UDP traffic, i.e., similar trends persist.

Robustness Study
The motivation for us to do a robustness study of different data collection methods is (i) it is not easy to get a good estimation of available bandwidth of a path and (ii) network congestion conditions might change over time, and hence it is important to see how sensitive the results are to the accuracy of available bandwidth information. To this end we use perturbed values of available bandwidth in computing the data transfer schedule for both non-coordinated and coordinated methods. That is, we deviate available bandwidth values from the actual estimates to emulate inaccuracies in available bandwidth estimates. Note that, the direct method does not use available bandwidth estimation information, and is therefore unaffected by the errors in the estimation.

Since more interesting scenarios correspond to inaccuracies in bandwidth estimation of congested points, (B0,A0) and (B1,A1), we consider the following two cases: (a) increasing the mis-estimates of the available bandwidth of paths passing through link (B0,A0) while decreasing the mis-estimates of the available bandwidth of paths passing through link (B1,A1), and (b) decreasing the mis-estimates of the available bandwidth of paths passing through link (B0,A0) while increasing the mis-estimates of the available bandwidth of paths passing through link (B1,A1). We ran a number of experiments corresponding to different loadings on peering points and different volumes of UDP traffic, and the results were qualitatively similar. Therefore, we only present an experiment with the ratio of traffic loads between peering point (B0,A0) and peering point(B1,A1) being 1:3. Figure 4 shows a series of results from over-estimating the available bandwidth of paths passing through the more congested link by 70% in (a), 50% in (b), and 30% in (c), to accurate estimation in (d), to under-estimating the available bandwidth of paths passing through the more congested link by 50% in (e), and 70% in (f). Although the coordinated method is affected by estimation errors in all paths, since it uses multiple paths to perform transfers, the effect is relatively small. For instance, even when the available bandwidth of the paths passing through the more congested link is over-estimated by 70%, as in Figure 4(a), the makespan is increased by 46% when comparing to the same method with accurate bandwidth estimation. If we under-estimate the available bandwidth by 70%, as in Figure 4(f), the makespan is increased by 24%. However, note that in all cases, the non-coordinated method performs worse than the coordinated method, anywhere from 37% to 142%. Furthermore, for the non-coordinated method, a mis-estimation can be more dangerous; e.g., a high over-estimation of the available bandwidth of a more congested link, as in Figures 4(a) and 4(b), can result in most of the traffic being routed through the more congested path. Thus, the resulting performance would degrade sharply and can be even worse than that of the direct method. As shown in Figure 4(a), the makespan of the non-coordinated method is more than double of the makespan of the same method with accurate bandwidth estimation. On the other hand, since the non-coordinated method chooses the best path (in term of available bandwidth), this over-estimation needs to be sufficiently high in order to force the wrong choice of best path. For instance, in Figure 4(c), it does not affect the makespan. We note that, under-estimating the available bandwidth of paths passing through a highly congested link does not affect the non-coordinated method since the method never picks that path.

Results of Throughput Metrics
Figure 5 shows how different methods affect the throughput of other network traffic[7]. Both the non-coordinated and the direct methods have a smaller impact on the throughput of the other network traffic than the coordinated method. However, the difference between the direct method and the coordinated method in our experiments is only 2% to 9%. This is not surprising since the coordinated method is more aggressive than other methods in taking advantage of bandwidth available in the network.

[7]We only consider TCP traffic since UDP traffic does not adjust its throughput when congestion occurs.

5 Conclusions

In this work, we studied the performance and robustness of different data transfer approaches to the large-scale data collection problem under various network congestion load and accuracies in estimation of bandwidth available. We showed that coordinated methods perform significantly better than non-coordinated and direct methods. Specifically, these improvements are achieved under low storage requirement overheads and without significant detrimental effects on other network traffic throughput. Moreover, we have showed that under mis-estimation situations coordinated methods still perform better than non-coordinated methods. Under high degree of mis-estimation, the performance of coordinated methods are less sensitive to such mis-estimation than the performance of non-coordinated methods. Therefore, we believe that coordinated methods are a promising approach to large-scale data collection problems.

References

[1] M. Allman. Measuring end-to-end bulk transfer capacity. *Proceedings of ACM SIGCOMM Internet Measurement Workshop*, 2001.

[2] David G. Andersen, Hari Balakrishnan, M. Frans Kaashoek, and Robert Morris. Resilient overlay networks. In *18th ACM SOSP Conference*, Canada, October 2001. ACM.

[3] B. R. Badrinath and P. Sudame. Gathercast: An efficient mechanism for multi-point to point aggregation in ip networks. Technical Report DCS-TR-362, Computer Science Department, Rutgers University, July 1998.

[4] S. Bhattacharjee, W. C. Cheng, C.-F. Chou, L. Golubchik, and S. Khuller. Bistro: a platform for building scalable wide-area upload applications. *ACM SIGMETRICS Performance Evaluation Review (also presented at the Workshop on Performance and Architecture of Web Servers (PAWS) in June 2000)*, 28(2):29–35, September 2000.

[5] K. Calvert, J. Griffioen, B. Mullins, A. Sehgal, and S. Wen. Concast: Design and Implementation of an Active Network Service. *IEEE Trans. on Networking*, 2000.

[6] W. C. Cheng, C.-F. Chou, L. Golubchik, S. Khuller, and Y. C. Wan. On a graph-theoretic approach to scheduling large-scale data transfers. Technical Report CS-TR-4322, University of Maryland, April 2002.

[7] C. Dovrolis and M. Jain. Pathload: A measurement tool for end-to-end available bandwidth. *Proceedings of the 3rd Passive and Active Measurements Workshop*, 2002.

[8] http://www-mash.cs.berkeley.edu/ns/. *UCB/LBNL/VINT Network Simulator - ns (version 2)*.

[9] http://www.cc.gatech.edu/fac/Ellen.Zegura/graphs.html. *Georgia Tech Internetwork Topology Generator*.

[10] IRS. *Fill-in Forms*. http://www.irs.ustreas.gov/prod/forms_pubs/fillin.html, 2001.

[11] T. Leighton. *The Challenges of Delivering Content on the Internet*. Keynote address at the ACM SIGMETRICS 2001 Conference, Cambridge, Massachusetts, June 2001.

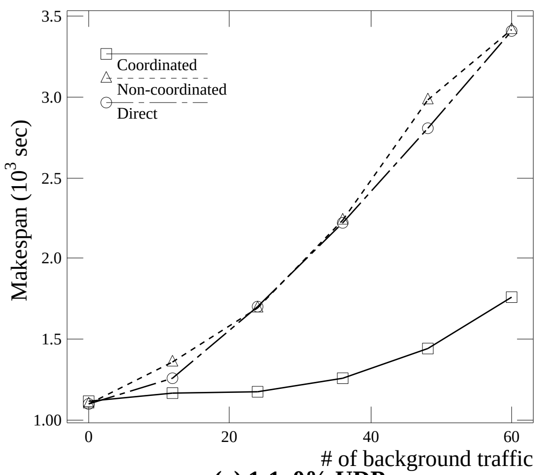
[12] M. Mathis and M. Allman. A framework for defining empirical bulk transfer capacity metrics. *RFC 3148*, 2001.

[13] Dan Rubenstein, James F. Kurose, and Donald F. Towsley. Detecting shared congestion of flows via end-to-end measurement. In *Proceedings of 2000 ACM SIGMETRICS Conf.*, pages 145–155, 2000.

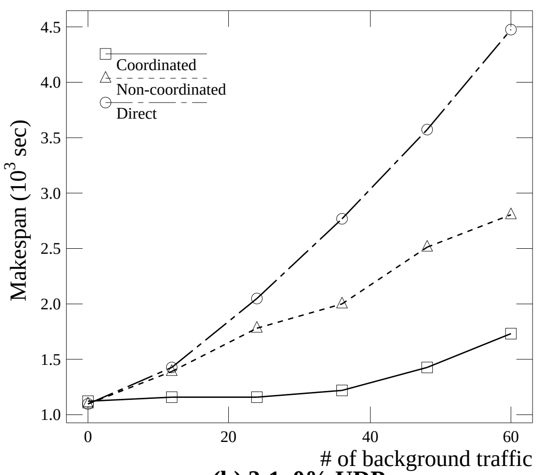
[14] Stefan Savage, Tom Anderson, Amit Aggarwal, David Becker, Neal Cardwell, Andy Collins, Eric Hoffman, John Snell, Amin Vahdat, Geoff Voelker, and John Zahorjan. Detour: a case for informed internet routing and transport. In *IEEE Micro*, volume 19, pages 50–59. IEEE, January 1999.

[15] Stefan Savage, Andy Collins, and Eric Hoffman. The end-to-end effects of internet path selection. In *ACM SIGCOMM '99 conference on Applications, technologies, architectures, and protocols for computer communication*, 1999.

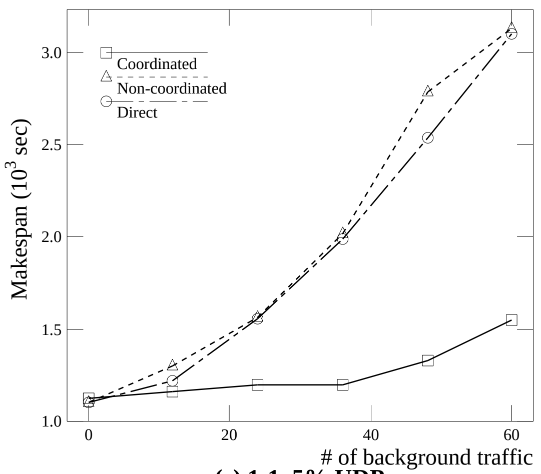
[16] B. Schneier. *Applied Cryptography, Second Edition*. Wiley, 1996.



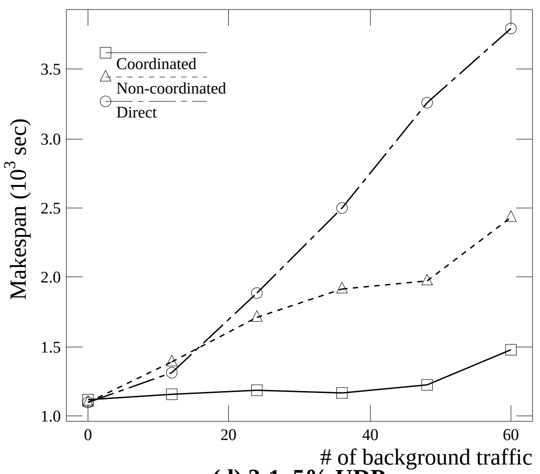
(a) 1:1, 0% UDP



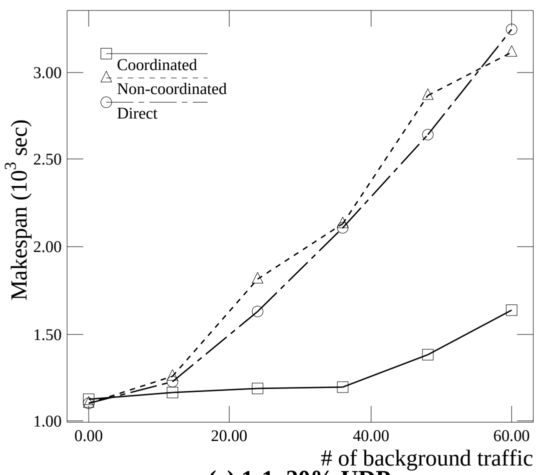
(b) 2:1, 0% UDP



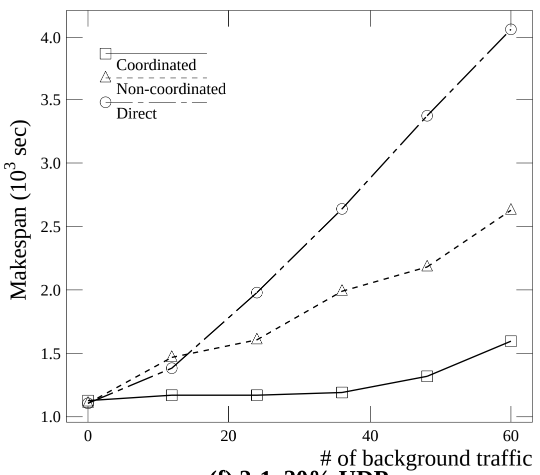
(c) 1:1, 5% UDP



(d) 2:1, 5% UDP

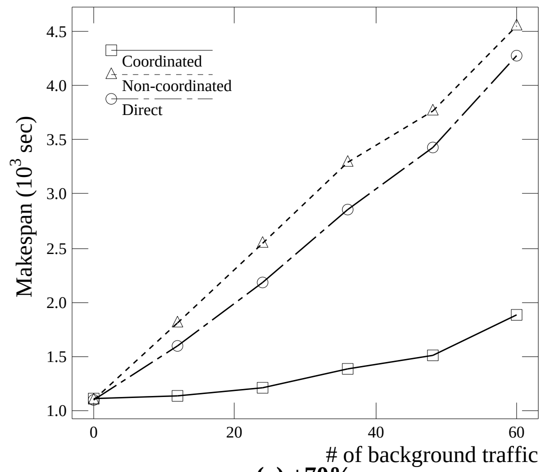


(e) 1:1, 20% UDP

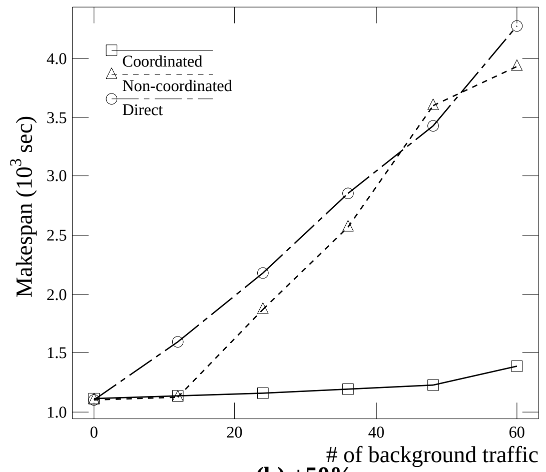


(f) 2:1, 20% UDP

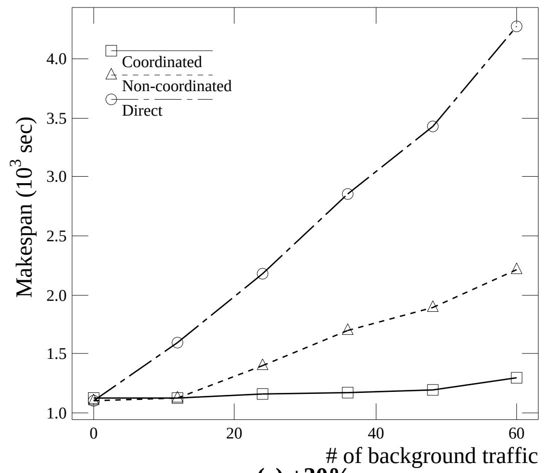
Figure 3: Makespan comparison under the effect of UDP traffic, and symmetric (1:1) and asymmetric (2:1) traffic.



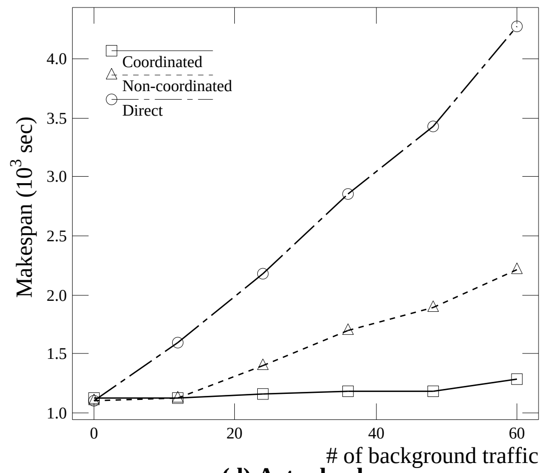
(a) +70%



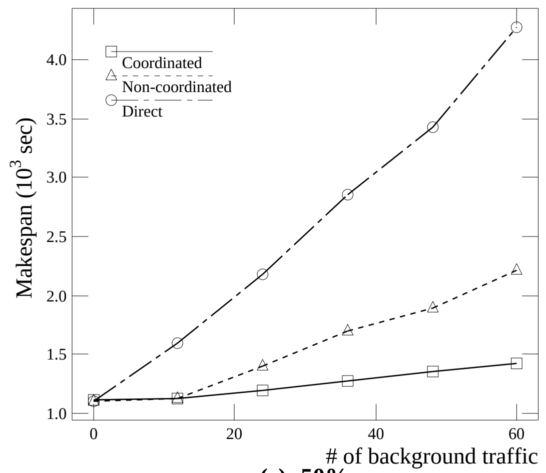
(b) +50%



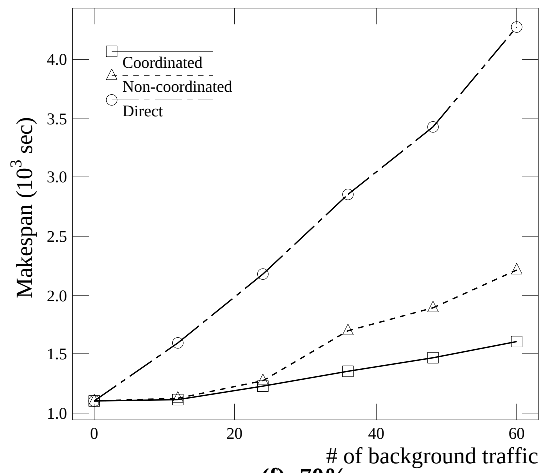
(c) +30%



(d) Actual value

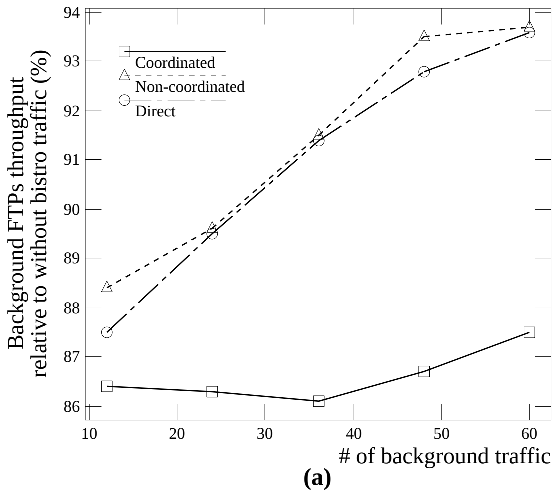


(e) -50%

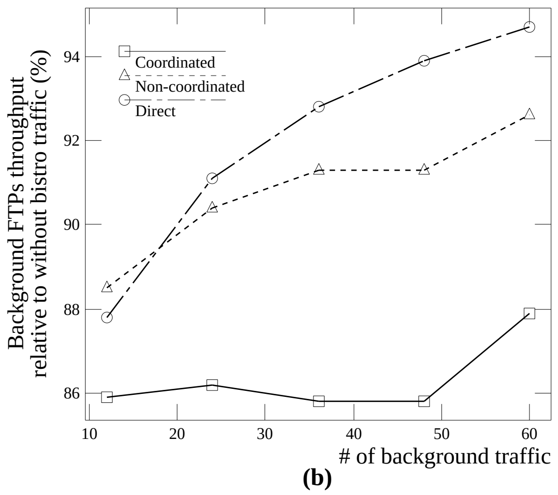


(f) -70%

Figure 4: Sensitivity to bandwidth estimation under 5% UDP traffic and ratio of loads on peering points of 1:3, under the Makespan Metric. Individual figure captions indicate the percentage of mis-estimation of paths going through (B1,A1).



(a)



(b)

Figure 5: Throughput comparsion under the effect of (a) symmetric (1:1) and (b) asymmetric (2:1) traffic.

Appendix A: Background on Bistro

This Appendix is included for the convenience of reviewers only; it is intended to be removed in the final version of the paper.

In this exposition we focus on upload applications with reasonably large data transfers and deadlines for clients to submit their data, e.g., such as the online income tax submission application mentioned in Section 1. However, we note that deadlines are not a factor in this work since the focus of this paper is on the performance of Step 3 below, which would typically be performed after the deadline. Hence, there is no deadline associated with this step, but its performance is still crucial since the data cannot be processed by the application until it is collected.

Briefly, the Bistro upload architecture works as follows (refer to [4] for details). Given a large number of clients that

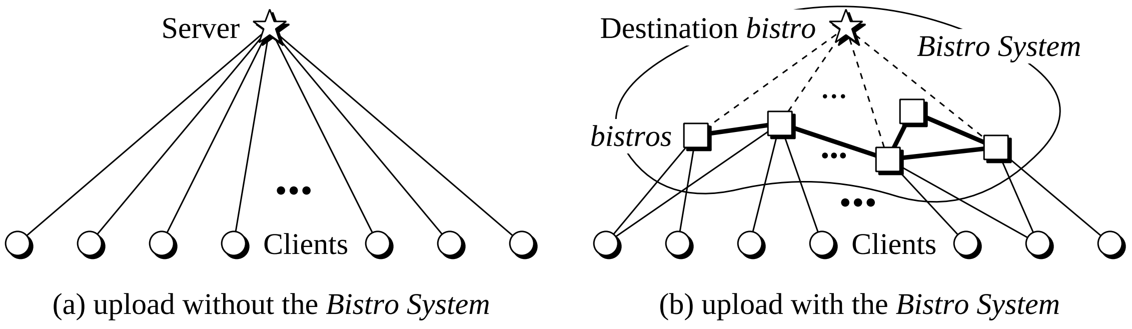

Server
Clients
(a) upload without the Bistro System
Destination bistro
Bistro System
bistros
Clients
(b) upload with the Bistro System

Figure 6: Upload Problem.

need to upload their data by a given deadline to a given destination server (refer to Figure 6(a)), the Bistro architecture breaks the upload problem into three steps (as illustrated in Figure 6(b)):

- *Step 1*, the timestamp step, which must be accomplished prior to the deadline for clients to submit their data to the destination server. In this step, each client sends to the server a message digest of their data [16] and in return receives a timestamp ticket from the destination server as a receipt indicating that the client made the deadline for data submission. The purpose of this step is to ensure that the client makes the deadline *without* having to transfer their data which is significantly larger than a message digest and might take a long time to transfer during high loads which are bound to occur around the deadline time. It is also intended to ensure that the client (or an intermediate bistro used in Step 2 below) does not change their data after receiving the timestamp ticket (hence the sending of the message digest to the destination server). All other steps can occur before or after the deadline.
- *Step 2*, the transfer of data from clients to intermediate hosts, termed bistros. This results in a low data transfer response time for clients since (a) the load of many clients is distributed among multiple bistros and (b) a good or near-by bistro can be selected for each client to improve data transfer performance. Since the bistros are not trusted entities (unlike the destination server), the data is encrypted by the client prior to the transfer.
- *Step 3*, the collection of data by the destination server from the bistros. The destination server determines when and how the data is collected in order to avoid hotspots around the destination server (i.e., the original problem of having many sources transfer their data to the same server around the same time). Once the destination server collects all the data, it can decrypt it, recompute message digests, and verify that no changes were made to a client's data (either by the client or by one of the intermediate bistros) after the timestamp ticket was issued.

A summary of main advantages of this architecture is as follows: (1) hotspots can be eliminated around the server because the transfer of data is decoupled from making of the deadline, (2) clients can receive good performance since they can be dispersed among many bistros and each one can be direct to the "best" bistro for that client, and (3) the destination server can minimize the amount of time it takes to collect all the data since now it is in control of when and how to do it (i.e., Bistro employs a server pull).

The performance evaluation of different approaches to Step 3, i.e., the large-scale data collection problem, are the focus of this paper.

Appendix B: Overview of Coordinated Method

This Appendix is included for the convenience of reviewers only; it is intended to be removed in the final version of the paper.

In this section, we give an overview of the coordinated approach (please refer to [6] for details) to the large-scale data collection problem stated in Section 1.

- *Step 1.* Construct an overlay graph representation of the hosts connected through the network. That is, each host is corresponding to a node in the graph, a path between two hosts corresponds to a link, and (estimated) available bandwidth of a path represents the capacity of the corresponding link.
- *Step 2.* Generate a time-expanded version of the overlay graph constructed in Step 1, i.e., this time-expanded graph is to consider discrete steps of time and make a copy of the original overlay graph at each time step. The intuition to do so is if the capacity of a link is x data units, we could use that link to transfer x units of data for each time step.
- *Step 3.* Determine a data transfer schedule on the time-expanded graph by optimizing a given objective function under the appropriate set of constraints. In this paper, we focus on the objective of minimizing the total amount of time it takes to collect the data from the source hosts, i.e., *makespan*.
- *Step 4.* Convert the solution produced in Step 3 under the graph theoretic formulation to a data transfer schedule over a communication network. This schedule must specify on what path and in what order should each "piece" of data be transferred to the destination host, where a path is defined as a sequence of hosts, with the first host on the path being the source of the data, intermediate hosts on the path being other hosts, and the last host on the path being the destination host.