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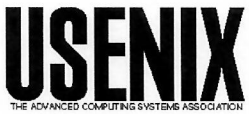

Designing a DHT for low latency and high throughput

Frank Dabek, Jinyang Li, Emil Sit, James Robertson, M. Frans Kaashoek, Robert Morris *
MIT Computer Science and Artificial Intelligence Laboratory
fdabek, jinyang, sit, jsr, kaashoek, rtm@csail.mit.edu

Abstract

Designing a wide-area distributed hash table (DHT) that provides high-throughput and low-latency network storage is a challenge. Existing systems have explored a range of solutions, including iterative routing, recursive routing, proximity routing and neighbor selection, erasure coding, replication, and server selection.

This paper explores the design of these techniques and their interaction in a complete system, drawing on the measured performance of a new DHT implementation and results from a simulator with an accurate Internet latency model. New techniques that resulted from this exploration include use of latency predictions based on synthetic coordinates, efficient integration of lookup routing and data fetching, and a congestion control mechanism suitable for fetching data striped over large numbers of servers.

Measurements with 425 server instances running on 150 PlanetLab and RON hosts show that the latency optimizations reduce the time required to locate and fetch data by a factor of two. The throughput optimizations result in a sustainable bulk read throughput related to the number of DHT hosts times the capacity of the slowest access link; with 150 selected PlanetLab hosts, the peak aggregate throughput over multiple clients is 12.8 megabytes per second.

1 Introduction

The Internet has transformed communication for distributed applications: each new system need not implement its own network, but can simply assume a shared global communication infrastructure. A similar transformation might be possible for storage, allowing distributed applications to assume a shared global storage infrastructure. Such an infrastructure would have to name and find data, assure high availability, balance load across available servers, and move data with high throughput and low latency.

Distributed hash tables (DHTs) are a promising path towards a global storage infrastructure, and have been used as the basis for a variety of wide-area file and content publishing systems [13, 26, 34, 38]. Good performance, however, is a challenge: the DHT nodes holding the data may be far away in the network, may have access link capacities that vary by orders of magnitude, and may experience varying degrees of congestion and packet loss.

This paper explores design choices for DHT read and write algorithms. Existing work has investigated how to make the *lookup of keys* in DHTs scalable, low-latency, fault-tolerant, and secure, but less attention has been paid to the efficiency and robustness with which DHTs *read and store data*. This paper considers a range of design options for efficient data handling in the context of a single DHT, DHash++. The decisions are evaluated in simulation and in an implementation of DHash++ on the PlanetLab [29] and RON [2] test-beds.

To bound the discussion of design decisions, we have made a number of assumptions. First, we assume that all nodes cooperate; the algorithms for reading and writing are likely to be more expensive if they have to defend against malicious nodes. Second, we assume that lookups are routed using one of the $O(\log N)$-style schemes, instead of using the recently proposed $O(1)$ schemes [14, 17, 18, 44]. Finally, we assume that the DHT stores small blocks (on the order of 8192 bytes). Relaxing these assumptions will result in different DHT designs with different latency and throughput properties, which we hope to explore in the future.

The paper makes the following contributions. Recursive lookups take about 0.6 times as long as iterative; the reason why the reduction is not a factor of two is the cost of the final return trip. The latency of the last few hops in a lookup acts as a lower bound on the performance of Proximity Neighbor Selection [37, 16], which approximates 1.5 times the average round trip time in the underlying network. This result holds regardless of the number of DHT nodes (and thus regardless of the number of hops). Replicated data allows for low-latency reads because there are many choices for server selection, while erasure-coded data reduces bandwidth consumption for writes at the expense of increased read latency. Integration of key lookup and data fetch reduces the lower bound imposed by the last few lookup hops. Finally, using an integrated trans-

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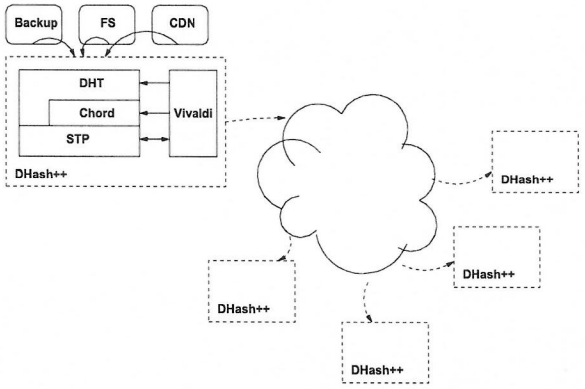

Figure 1: DHash++ system overview.

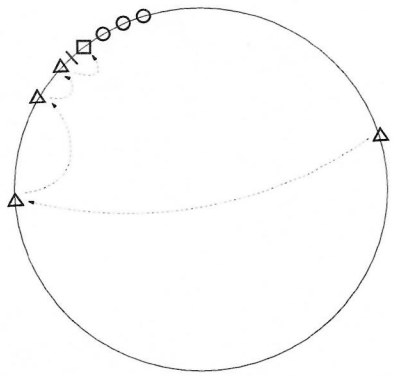

Figure 2: An illustration of a Chord identifier ring. The tick mark denotes the position of a key in ID space. The square shows the key's successor node, and the circles show the nodes in the successor's successor list. The triangles and arrows show a lookup path. The last node before the tick mark is the key's predecessor.

port protocol rather than TCP provides opportunities for efficiency in alternate routing after timeouts and allows the DHT freedom to efficiently contact many nodes.

The rest of this paper is structured as follows. Section 2 outlines the complete system that surrounds the specific mechanisms detailed in the paper. Section 3 describes the methods behind the paper's measurements and quantitative evaluations. Section 4 discusses design decisions that affect latency, and Section 5 discusses throughput. Section 6 describes related work. We conclude in Section 7.

2 Background

For concreteness, this evaluates design decisions in the context of a complete DHT called DHash++. This section describes the parts of DHash++ that are needed to understand the rest of the paper.

2.1 Chord

DHash++ uses the Chord lookup algorithm to help it find data [42]. Chord provides a function `lookup(key)` $\rightarrow$ `set-of-IP`, which maps a 160-bit key to the set of IP addresses of the nodes responsible for that key. Each node has a 160-bit identifier, and Chord designates the s nodes whose identifiers immediately follow a key as responsible for that key; these are the key's *successors*. To provide reliable lookup even if half of the nodes fail in a 2^{16}-node network, the number of successors, s, is 16 in the Chord implementation. The ID space wraps around, so that zero immediately follows $2^{160} - 1$.

The *base Chord* lookup algorithm (which will be modified in subsequent sections) works as follows. Each Chord node maintains a *finger table*, consisting of the IP addresses and IDs of nodes that follow it at power-of-two distances in the identifier space. Each node also maintains a *successor list* referring to its s immediate successors. When a node originates a lookup, it consults a sequence of other nodes, asking each in turn which node to talk to next. Each node in this sequence answers with the node from its finger table with highest ID still less than the desired key. The originating node will find the key's *predecessor* node after $O(\log N)$ consultations; it then asks the predecessor for its successor list, which is the result of the lookup. This style of lookup is called *iterative*, since the originating node controls each step of the lookup. All of the communication uses UDP RPCs.

Figure 2 shows a Chord ring with a key, its successor, the successor's successor list, and a lookup path; this picture is helpful to keep in mind since much of the discussion appeals to the ring geometry. Although this paper explores optimizations over base Chord, we believe that these optimizations also apply to other DHTs that route in ID spaces using an $O(\log N)$ protocol.

2.2 DHash++

DHash++ stores key/value pairs (called blocks) on a set of servers. The DHash++ client API consists of `key` $\leftarrow$ `put(value)` and `get(key)` $\rightarrow$ `value`. DHash++ calculates the key to be the SHA-1 hash of the value, and uses Chord to decide which server should store a given block; each server runs both Chord and DHash++ software. As well as finding and moving data for client applications, DHash++ authenticates the data and moves it from server to server as nodes join, leave, and fail [7].

2.3 Synthetic coordinates

Many of the techniques described in this paper use synthetic coordinates to predict inter-node latencies without having to perform an explicit measurement to determine the latency. A number of synthetic coordinate systems have been proposed [10, 24, 27, 30, 33, 39]. We chose

to use Vivaldi [12], because its algorithm is decentralized, which makes it suitable for use in peer-to-peer systems. Furthermore, the Vivaldi algorithm is lightweight, since it can piggy-back on DHash++'s communication patterns to compute coordinates.

Whenever one Chord or DHash++ node communicates directly with another, they exchange Vivaldi coordinates. Nodes store these coordinates along with IP addresses in routing tables and successor lists. The result of a lookup for a key carries the coordinates of the nodes responsible for the key as well as their IP addresses. Thus the requesting node can predict the latencies to each of the responsible nodes without having to first communicate with them.

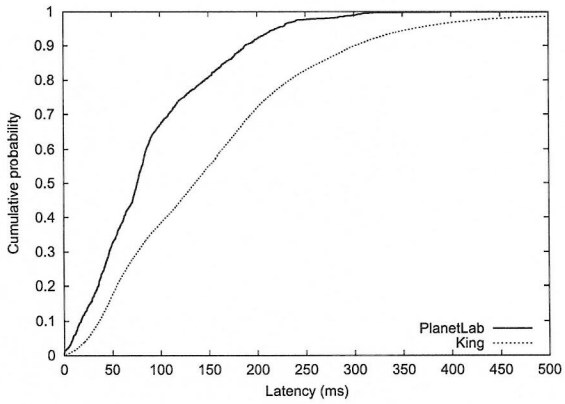

Figure 3: Round-trip latency distribution over all pairs of PlanetLab and King dataset hosts. The median and average King dataset latencies are 134 and 154 milliseconds respectively. The median and average PlanetLab latencies are 76 and 90 milliseconds respectively.

3 Evaluation methods

The results in this paper are obtained through simulations and measurements on the PlanetLab and RON test-beds. The measurements focus on DHT operations that require low latency or high throughput.

3.1 Evaluation infrastructure

DHT performance depends on the detailed behavior of the servers and the underlying network. The test-bed measurements in Section 4 were taken from a DHash++ implementation deployed on the PlanetLab and RON test-beds. 180 test-bed hosts were used, of which 150 were in the United States and 30 elsewhere. 105 of the hosts are on the Internet2 network; the rest have connections via DSL, cable modem, commercial T1 service, or are at co-location centers. Each host runs three independent DHash++ processes, or *virtual nodes*, in order to improve load balance and to ensure that the total number of nodes is large compared to the size of the Chord successor list. The measurements in Section 5 were taken on the 27-node RON test-bed alone.

The test-bed measurements are augmented with simulation results to explore large configurations, to allow easy testing of alternate designs, and to allow analytic explanations of behavior in a controlled environment. The simulated network models only packet delay. One input to the simulator is a full matrix of the round-trip delays between each pair of simulated hosts. This approach avoids having to simulate the Internet's topology, a currently open area of research; it requires only the measurement of actual pair-wise delays among a set of hosts. The simulator can produce useful speed-of-light delay results, but cannot be used to predict throughput or queuing delay.

The simulator's delay matrix is derived from Internet measurements using techniques similar to those described by Gummadi et al. [15]. The measurements involved 2048 DNS servers found with inverse DNS lookups on a trace of over 20,000 Gnutella clients. For each pair of these servers, a measuring node sends a query to one server that requires it to contact the other server. Subtracting the delay between the measuring node and the first server from the total delay yields the delay between the two servers. In order to reduce the effects of queuing delay, the minimum delay from five experiments is used. In this paper the results are called the King data-set. All the simulations in this paper involve 2048 DHT nodes using King delay matrix unless otherwise mentioned. Figure 3 shows the CDF of the King data-set round-trip times; the median is 134 milliseconds, while the average is 154 milliseconds. The graph also shows the minimum delay of five pings between each pair of PlanetLab hosts for comparison. The main difference between the two curves is the longer tail on the King distribution, which is likely caused by the larger sample of nodes.

3.2 Application workload

The design of a DHT must incorporate assumptions about probable application behavior, and a DHT evaluation must also involve either applications or models of application behavior. The application aspects that most affect performance are the mix of read and write operations, the degree to which operations can be pipelined, and the size of the data records.

DHash++ is designed to support read-heavy applications that demand low-latency and high-throughput reads as well as reasonably high-throughput writes. Examples of such applications might include the Semantic Free Referencing system (SFR) [45] and UsenetDHT [40].

SFR is a naming system designed to replace the use of DNS as a content location system. SFR uses a DHT to store small data records representing name bindings.

Reads are frequent and should complete with low latency. Writes are relatively infrequent and thus need not be as high performance. SFR data blocks are likely to be on the order hundreds of bytes.

UsenetDHT is a service aiming to reduce the total storage dedicated to Usenet by storing all Usenet articles in a shared DHT. UsenetDHT splits large binary articles (averaging 100 KB) into small blocks for load balance, but smaller text articles (typically 5 KB or less) are stored as single blocks. While readership patterns vary, UsenetDHT must support low-latency single article reads, as well as high-throughput pipelined article fetches.

These systems are unlikely to be deployed on high-churn networks—these systems are all server-class. The target environment for them is a network with relatively reliable nodes that have good Internet access.

4 Designing for low latency

This section investigates five design choices that affect DHT `get` latency. The naive algorithm against which these choices are judged, called *base DHash++*, operates as follows. Each 8192-byte block is stored as 14 1171-byte erasure-coded fragments, any seven of which are sufficient to reconstruct the block, using the IDA coding algorithm [31]. The 14 fragments are stored at the 14 immediate successors of the block's key. When an application calls `get(key)`, the originating node performs an iterative Chord lookup, which ends when the key's predecessor node returns the key's 16 successors; the originating node then sends seven parallel requests the first seven successors asking them each to return one fragment.

Figure 4 gives a preview of the results of this section. Each pair of bars shows the median time to fetch a block on the PlanetLab test-bed after cumulatively applying each design improvement. The design improvements shown are recursive rather than iterative routing, proximity neighbor selection, fetching of data from the closest copy, and integration of lookup routing and data fetching. These design improvements together reduce the total fetch latency by nearly a factor of two.

This paper uses a $log(N)$ protocol for routing lookups. An optimization that isn't explored in this paper is an increase in the base to reduce the number of hops, or the use of a constant-hop protocols. These optimizations would reduce latency under low churn, because each node would know about many other nodes. On the other hand, in high churn networks, these optimizations might require more bandwidth to keep routing tables up to date or experience more timeouts because routing tables might contain recently-failed nodes. The paper's evaluation infrastructure isn't adequate to explore this design decision in detail. We hope to explore this issue in future work. We do explore the extent to which proximity routing can reduce the impact of the number of hops on the lookup latency.

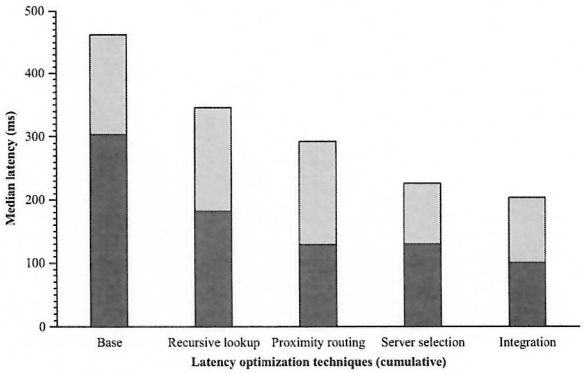

Figure 4: The cumulative effect of successive optimizations on the latency of a DHash++ data fetch. Each bar shows the median time of 1,000 fetches of a randomly chosen 8192-byte data block from a randomly chosen host. The dark portion of each bar shows the lookup time, and the light portion shows the time taken to fetch the data. These data are from the implementation running on PlanetLab.

4.1 Data layout

The first decision to be made about where a DHT should store data is whether it should store data at all. A number of DHTs provide only a key location service, perhaps with a layer of indirection, and let each application decide where (or even whether) to store data [20, 28]. The choice is a question of appropriate functionality rather than performance, though Section 4.5 describes some performance benefits of integrating the DHT lookup and data storage functions. The approach taken by DHash++ is appropriate for applications that wish to view the DHT as a network storage system, such as our motivating examples SFR and UsenetDHT.

For DHTs that store data, a second layout decision is the size of the units of data to store. A DHT key could refer to a disk-sector-like block of data [13], to a complete file [38], or to an entire file system image [11]. Large values reduce the amortized cost of each DHT lookup. Small blocks spread the load of serving popular large files. For these reasons, and because some applications such as SFR require the DHT to store small blocks, DHash++ is optimized with blocks of 8 KB or less in mind.

A third layout decision is which server should store each block of data (or each replica or coded fragment). If a given block is likely to be read mostly by hosts in a particular geographic area, then it would make sense to store the data on DHT servers in that area. Caching is one way to achieve this kind of layout. On the other hand, geographic concentration may make the data more

vulnerable to network and power failures, it may cause the load to be less evenly balanced across all nodes, and is difficult to arrange in general without application hints. At the other extreme, the DHT could distribute data uniformly at random over the available servers; this design would be reasonable if there were no predictable geographic locality in the originators of requests for the data, or if fault-tolerance were important. DHash++ uses the latter approach: a block's key is essentially random (the SHA-1 of the block's value), node IDs are random, and a block's replicas or fragments are placed at its key's successor nodes. The result is that blocks (and load) are uniformly spread over the DHT nodes, and that a block's replicas or fragments are widely scattered to avoid correlated failure.

Given a DHT design that stores blocks on randomly chosen servers, one can begin to form some expectations about fetch latency. The lower bound on the total time to find and fetch a block is the round trip time from the originator to the nearest replica of the block, or the time to the most distant of the closest set of fragments required to reconstruct the block. For the typical block this time is determined by the distribution of inter-host delays in the Internet, and by the number of choices of replicas or fragments. The DHT lookup required to find the replicas or fragments will add to this lower bound, as will mistakes in predicting which replica or fragments are closest.

Most of the design choices described in subsequent subsections have to do with taking intelligent advantage of choices in order to reduce lookup and data fetch latency.

4.2 Recursive or iterative?

The base Chord and Kademlia algorithms are iterative: the originator sends an RPC to each successive node in the lookup path, and waits for the response before proceeding [25, 42]. Another possibility is recursive lookup [6, 47]: each node in the lookup path directly forwards the query to the next node, and when the query reaches the key's predecessor, the predecessor sends its successor list directly back to the originator [42]. Recursive lookup, which many DHTs use, might eliminate half the latency of each hop since each intermediate node can immediately forward the lookup before acknowledging the previous hop.

Figure 5 shows the effect of using recursive rather than iterative lookup in the simulator with the 2048-node King data set. For each technique, 20,000 lookups were performed, each from a random host for a random key. The average number of hops is 6.3. Recursive lookup takes on average 0.6 times as long as iterative. This decrease is not quite the expected factor of two: the difference is due to the extra one-way hop of (on average) 77 milliseconds to return the result to the originator.

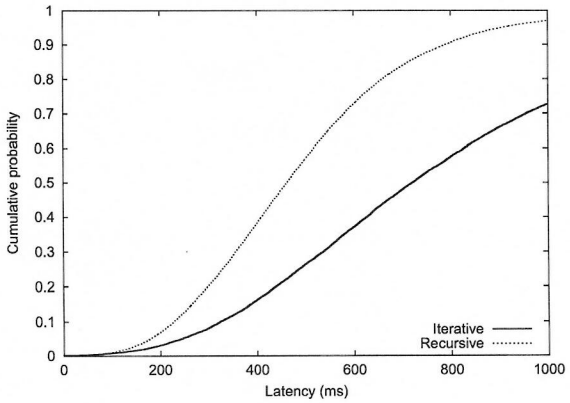

Figure 5: The cumulative distributions of lookup time for Chord with recursive and iterative lookup. The recursive median and average are 461 and 489 milliseconds; the iterative median and average are 720 and 822 milliseconds. The numbers are from simulations.

While recursive lookup has lower latency than iterative, iterative is much easier for a client to manage. If a recursive lookup elicits no response, the originator has no information about what went wrong and how to re-try in a way that is more likely to succeed. Sometimes a simple re-try may work, as in the case of lost packets. If the problem is that each successive node can talk to the next node, but that Internet routing anomalies prevent the last node from replying to the originator, then re-tries won't work because only the originator realizes a problem exists. In contrast, the originator knows which hop of an iterative lookup failed to respond, and can re-try that hop through a different node in the same region of the identifier space.

On the the other hand, recursive communication may make congestion control easier (that is, it is it may make it more feasible to rely on TCP). We will show in Section 5 that the performance of a naive TCP transport can be quite poor.

DHash++ uses recursive lookups by default since they are faster, but falls back on iterative lookups after persistent failures.

4.3 Proximity neighbor selection

Many DHTs decrease lookup latency by choosing nearby nodes as routing table entries [6, 16, 25, 42, 43, 47], a technique often called proximity neighbor selection (PNS). The reason this is possible is that there are usually few constraints in the choice of routing entries: any node in the relevant portion of the identifier space is eligible. A DHT design must include an algorithm to search for nearby nodes; an exhaustive search may improve lookup

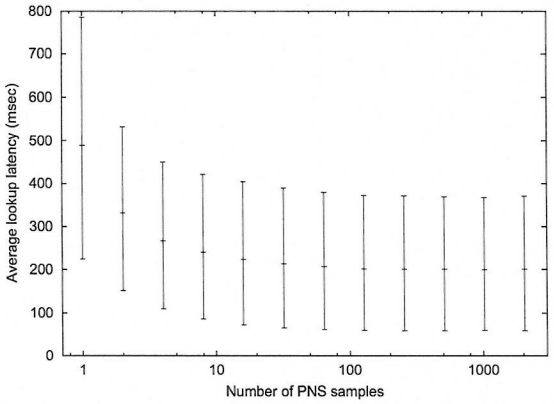

Figure 6: Average lookup latency as a function of the number of PNS samples. The bar at each x value shows the 10th, average, and 90th percentile of the latencies observed by 20,000 recursive lookups of random keys from random nodes using PNS(x). The measurements are from the simulator with 2048 nodes.

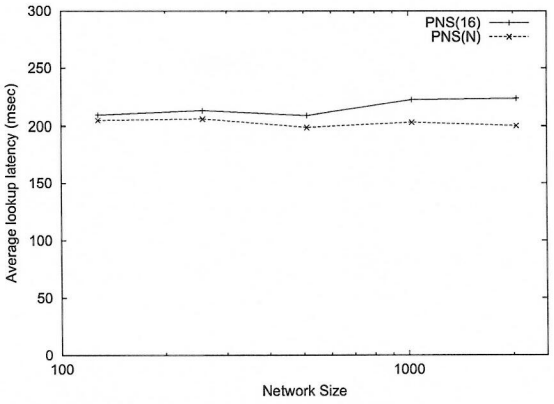

Figure 7: Average lookup latency of PNS(16) and PNS(N) as a function of the number of nodes in the system, N. The simulated network sizes consist of 128, 256, 512, 1024, 2048 nodes.

latency, but also consume network resources. This subsection builds on the work of Gummadi et al. [16] in two ways: it explains why PNS approximates 1.5 times the average round trip time in the underlying network and shows that this result holds regardless of the number of DHT nodes (and thus regardless of the number of hops).

Following Gummadi et al. [16], define PNS(x) as follows. The ith Chord finger table entry of the node with ID a properly refers to the first node in the ID-space range $a + 2^i$ to $a + 2^{i+1} - 1$. The PNS(x) algorithm considers up to the first x nodes in that range (there may be fewer than x), and routes lookups through the node with lowest latency. *Ideal PNS* refers to PNS(x) with x equal to the total number of nodes, so that every finger table entry points to the lowest-latency node in the entire allowed ID-space range. The simulator simply chooses the lowest-latency of the x nodes, while the real implementation asks each proper finger entry for its successor list and uses Vivaldi to select the closest node. This means that the real implementation requires that $x \leq s$ (the number of successors).

What is a suitable value for x in PNS(x)? Figure 6 shows the simulated effect of varying x on lookup latency. For each x value, 20,000 lookups were issued by randomly selected hosts for random keys. Each lookup is recursive, goes to the key's predecessor node (but not successor), and then directly back to the originator. The graph plots the median, 10th percentile, and 90th percentile of latency.

Figure 6 shows that PNS(1) has a simulated average latency of 489 ms, PNS(16) has an average latency of 224 ms, and PNS(2048) has an average latency of 201 ms. The latter is ideal PNS, since the neighbor choice is over all nodes in the simulation. PNS(16) comes relatively close to the ideal, and is convenient to implement in the real system with successor lists.

Why does ideal PNS show the particular improvement that it does? The return trip from the predecessor to the originator has the same median as the one-way delay distribution of the nodes in the network, δ. For the King data set, $\delta = 67$ms. The last hop (to the predecessor) has only one candidate, so its median latency is also δ. Each preceding hop has twice as many candidate nodes to choose from on average, since the finger-table interval involved is twice as large in ID space. So the second-to-last hop is the smaller of two randomly chosen latencies, the third-to-last is the smallest of four, etc. The minimum of x samples has its median at the $1 - 0.5^{\frac{1}{x}}$ percentile of the original distribution, which can be approximated as the $\frac{1}{x}$ percentile for large x. Doubling the sample size x will halve the percentile of the best sample. Assuming a uniform latency distribution, doubling the sample size halves the best sampled latency. Therefore, the latencies incurred at successive lookup hops with ideal PNS can be approximated by a geometric series with the final lookup hop to the key's predecessor being the longest hop. The lookup process includes an additional final hop to the originator. If we use the per-hop median latency as a gross approximation of the average per-hop latency, the total average lookup latency is thus approximated as: $\delta + (\delta + \frac{\delta}{2} + \frac{\delta}{4} + ...) = \delta + 2\delta = 3\delta$. For the King data set, this gives 201 ms. This is coincidentally the ideal PNS simulation result of 201 ms.

The fact that the average lookup latency of PNS(N) can be approximated as an infinite geometric series whose

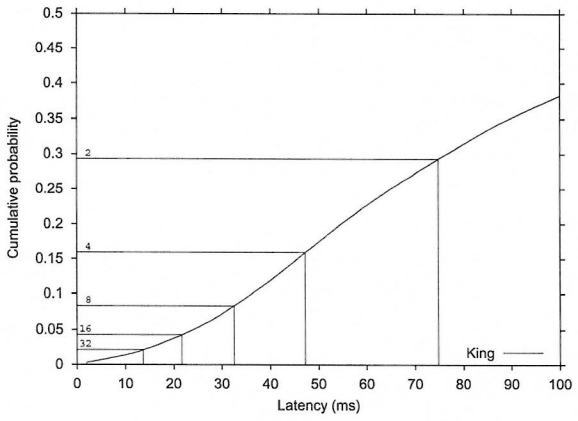

Figure 8: The median of the minimum latency taken from x samples out of the all-pairs empirical latency distribution of the King dataset. The boxes correspond to 2,4,8,16,32 samples starting from the right.

sum converges quickly suggests that despite the fact that the number of lookup hops scales as $log(N)$, the total average lookup latency will stay close to 3δ. Figure 7 shows the simulated average lookup latency as a function of the number of nodes in the system. As we can see, there is indeed little increase in average lookup latency as the network grows.

Why are there diminishing returns in Figure 6 beyond roughly PNS(16)? First, the King delay distribution is not uniform, but has a flat toe. Thus increasing the number of samples produces smaller and smaller decreases in minimum latency. Figure 8 shows this effect for various sample sizes. Second, for large x, the number of samples is often limited by the allowed ID-space range for the finger in question, rather than by x; this effect is more important in the later hops of a lookup.

One lesson from this analysis is that the last few hops of a lookup dominate the total latency. As a lookup gets close to the target key in ID space, the number of remaining nodes that are closer in ID space to the key decreases, and thus the latency to the nearest one increases on average. Section 4.5 shows how to avoid this problem.

4.4 Coding versus replication

Once the node originating a fetch acquires the key's predecessor's successor list, it knows which nodes hold the block's replicas [13, 38] or fragments of an erasure-coded block [8, 3, 22, 19]. In the case of replication, the originator's strategy should be to fetch the required data from the successor with lowest latency. The originator has more options in the case of coded fragments, but a reasonable approach is to fetch the minimum required number of fragments from the closest successors. The technique of fetching the data from the nearest of a set of candidate nodes is typically called server selection.

The design choice here can be framed as choosing the coding parameters l and m, where l is the total number of fragments stored on successors and m is the number required to reconstruct the block. Replication is the special case in which $m = 1$, and l is the number of replicas. The *rate* of coding, $r = \frac{l}{m}$, expresses the amount of redundancy. A replication scheme with three replicas has $m = 1, l = 3$, and $r = 3$, while a 7-out-of-14 IDA coding scheme has $m = 7, l = 14$, and $r = 2$.

The choice of parameters m and l has three main effects. First, it determines a block's availability when nodes fail [46]. If the probability that any given DHT node is available is p_0, the probability that a block is still available is [4]:

$$p_{avail} = \sum_{i=m}^{l} \binom{l}{i} p_0^i (1 - p_0)^{l-i} \quad (1)$$

Second, increasing r is likely to decrease fetch latency, since that provides the originator more choices from which to pick a nearby node. Third, increasing r increases the amount of communication required to write a block to the DHT. These performance aspects of erasure coding have not been considered previously.

Figure 9 illustrates the relationship between total fetch latency and block availability. The probability p_0 that each node is available is kept constant at 0.9. Each line represents a different rate r, and the points on the line are obtained by varying m and setting $l = r \times m$. Each point's x-axis value indicates the probability that a block is available as calculated by Equation 1. Each point's y-axis value is the average latency from 20,000 simulations of fetching a random block from a random originating node. The originator performs a lookup to obtain the list of the desired key's successors, then issues parallel RPCs to the m of those successors that have lowest latency, and waits for the last of the RPCs to complete. The y-axis values include only the data fetch time.

The left-most point on each line corresponds to replication; that point on the different lines corresponds to 2, 3, and 4 replicas. For each line, the points farther to the right indicate coding schemes in which smaller-sized fragments are placed onto larger numbers of nodes. For each redundancy rate r, replication provides the lowest latency by a small margin. The reason is easiest to see for $r = 2$: choosing the nearest k of $2k$ fragments approaches the median as k grows, while choosing the nearest replica of two yields a latency considerably below the median. Replication also provides the least availability because the redundant information is spread over fewer nodes. The lower lines correspond to larger amounts of redundant information on more nodes; this provides a wider

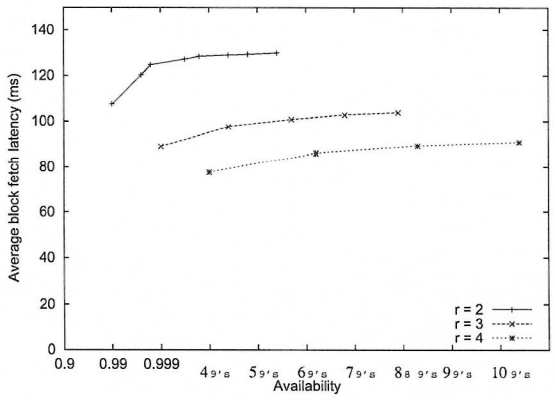

Figure 9: The relationship between read latency and block availability. The different lines correspond to different redundancy factors of $r = 2, 3, 4$. The data points on each line (starting from the left) correspond to reading $m = 1, 2, 3...$ fragments out of $r \times m$ total fragments. The x-axis value is computed from Equation 1 with the per-node availability p_0 set to 0.9, while the y-axis value is the simulated block fetch time (not including lookup time).

choice of nodes from which the originator can read the data, which increases the probability that it can read from nearby nodes, and lowers the fetch latency.

The best trade-off between replication and coding is dependent on the workload: a read-intensive workload will experience lower latency with replication, while a write-intensive workload will consume less network bandwidth with coding. DHash++ uses IDA coding with $m = 7$ and $l = 14$. The number seven is selected so that a fragment for an 8 KB block will fit in a single 1500-byte packet, which is important for UDP-based transport. The originator uses Vivaldi (Section 2.3) to predict the latency to the successors.

4.5 Integrating routing and fetching

So far the design of the DHT lookup algorithm and the design of the final data server-selection have been considered separately. One problem with this approach is that obtaining the complete list of a key's s successors requires that the originator contact the key's predecessor, which Section 4.3 observed was expensive because the final lookup steps can take little advantage of proximity routing. However, of the s successors, only the first l immediate successors store the fragments for the key's data block. Furthermore, fragments from any m of these successors are sufficient to reconstruct the block. Each of the $s - m$ predecessor nodes of the key has a successor list that contains m successors. Thus the lookup could stop

$a.\mathbf{lookup}(q, k, d)$:
  $overlap = \{n' \mid n' \in \text{succlist}_a \wedge n' > k\}$
  if $|overlap| \geq d$ **then**
    return $overlap$ to the originator q
  else if $overlap \neq \emptyset$ **then**
    $t = \{$the $s - d$ nodes in succlist_a immediately preceding $k\} \cup overlap$
    $b = t_i \in t$ s.t. $\text{dist}(a, t_i)$ is minimized
    if $b \in overlap$ **then**
      $t = b$.get succlist()
      $u =$ merger of t and $overlap$ to produce k first d successors
      return u to the originator q
    else
      return $b.\mathbf{lookup}(q, k, d)$
  else
    $b = \text{closestpred}(\text{lookupfinger}, k)$
    return $b.\mathbf{lookup}(q, k, d)$

Figure 10: Recursive lookup that returns at least d fragments of key k to sender q. Each node's successor list contains s nodes.

early at any of those predecessors, avoiding the expensive hop to the predecessor; Pastry/PAST uses a similar technique [38].

However, this design choice decreases the lookup time at the expense of data fetch latency, since it decreases the number of successors (and thus fragments) that the originator can choose from. Once the recursive lookup has reached a node n_1 whose successor list overlaps the key, n_1 is close enough to be the penultimate hop in the routing. By forwarding the query to the closest node n_2 in its successor list that can return enough nodes, n_1 can ensure that the next hop will be the last hop. There are two cases — if n_2 is past the key, then n_1 must directly retrieve n_2's successor list and merge it with its own overlapping nodes to avoid overshooting. Otherwise, n_1 can simply hand-off the query to n_2 who will have enough information to complete the request.

Figure 10 shows the pseudo-code for this final version of the DHash++ lookup algorithm. The d argument indicates how many successors the caller would like. d must be at least as large as m, while setting d to l retrieves the locations of all fragments.

The final latency design decision is the choice of d. A large value forces the lookup to take more hops, but yields more choice for the data fetch and thus lower fetch latency; while a small d lets the lookup finish sooner but yields higher fetch latency. Figure 11 explores this trade-off. It turns out that the cost of a higher d is low, since the lookup algorithm in Figure 10 uses only nearby nodes as the final hops, while the decrease in fetch time by using

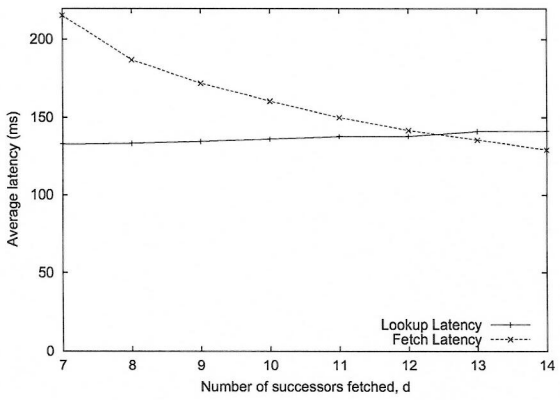

Figure 11: Simulated lookup and fetch time as a function of the d parameter in Figure 10. Larger d causes the lookup to take more hops and gather more successors; the extra successors decrease the fetch latency by providing more choice of nodes to fetch from. For comparison, the average lookup and fetch times that result from always contacting the predecessor are 224 and 129 milliseconds, respectively.

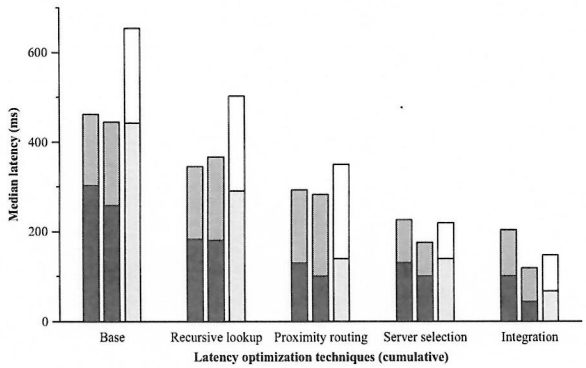

Figure 12: The cumulative effect of successive performance optimizations on the median latency of a DHash++ data fetch. The leftmost bar in each triple shows the time on our PlanetLab implementation (copied from Figure 4). The middle bar was produced by the simulator using a latency matrix measured between PlanetLab hosts. The rightmost bar corresponds to simulations of 2048 nodes using the King latency matrix. The dark portion of each bar shows the lookup time, and the light portion shows the time taken to fetch the data.

larger d is relatively large. Thus setting $d = l$ is the best policy.

4.6 Summary

Figure 12 summarizes the cumulative effect of the design decisions explored in this section. The leftmost bar in each triple shows the median time on our PlanetLab implementation (copied from Figure 4). The middle bar was produced by the simulator using a latency matrix measured between PlanetLab hosts. The dark portion of each bar shows the lookup time, and the light portion shows the time taken to fetch the data. Although the simulator results do not match the PlanetLab results exactly, the trends are the same. The results differ because the simulator uses inter-host delays measured between a slightly different set of PlanetLab nodes than were used for the implementation experiments, and at a different time.

The rightmost bar corresponds to simulations of 2048 nodes using the King latency matrix. The absolute numbers are larger than for the PlanetLab results, and perhaps more representative of the Internet as a whole, because the King data set includes a larger and more diverse set of nodes. Again, the overall trends are the same.

5 Achieving high throughput

Some applications, such as Usenet article storage, need to store or retrieve large amounts of data in a DHT. If data movement is to be fast, the DHT must make efficient use of the underlying network resources. The DHT must keep enough data in flight to cover the network's delay-bandwidth product, stripe data over multiple slow access links in parallel, and recover in a timely fashion from packet loss. The DHT must also provide congestion control in order to avoid unnecessary re-transmissions and to avoid overflowing queues and forcing packet loss. These goals are similar to those of traditional unicast transport protocols such as TCP [21], but with the additional requirement that the solution function well when the data is spread over a large set of servers.

This section presents two different designs, then compares their efficiency when implemented in DHash++ on the RON test-bed. We focus here on bulk fetch operations rather than insert operations.

5.1 TCP transport

Perhaps the simplest way for a DHT to manage its consumption of network resources is to use TCP. Because TCP imposes a start-up latency, requires time to acquire good timeout and congestion window size estimates, and consumes host state that limits the number of simultaneous connections, it makes the most sense for a DHT to maintain a relatively small number of long-running TCP connections to its neighbors and to arrange that communication only occur between neighbors in the DHT overlay. This arrangement provides congestion control without burdening the DHT with its implementation. Several systems use this approach (e.g., [34]), some with slight modifications to avoid exhausting the number of

file descriptors. For example, Tapestry uses a user-level re-implementation of TCP without in-order delivery [47].

Restricting communication to the overlay links means that all lookups and data movement must be recursive: iterative lookups or direct movement of data would not be able to use the persistent inter-neighbor TCP connections. Section 4.2 showed that recursive lookups work well. However, recursive data movement requires that each block of data be returned through the overlay rather than directly. This recursive return of data causes it to be sent into and out of each hop's Internet access link, potentially increasing latency and decreasing useful throughput. In addition, hiding the congestion control inside TCP limits the options for the design of the DHT's failure recovery algorithms, as well as making it hard for the DHT to control its overall use of network resources. Section 5.3 shows performance results that may help in deciding whether the convenience of delegating congestion control to TCP outweighs the potential problems.

DHash++ allows the option to use TCP as the transport. Each node keeps a TCP connection open to each of its fingers, as well as a connection to each node in its successor list. DHash++ forwards a get request recursively through neighbors' TCP connections until the request reaches a node whose successor list includes a sufficient number of fragments (as in Section 4.5). That node fetches fragments in parallel over the connections to its successors, trying the most proximate successors first. It then re-constructs the block from the fragments and sends the block back through the reverse of the route that the request followed. Pond [34] moves data through the Tapestry overlay in this way.

5.2 STP transport

At the other extreme, a DHT could include its own specialized transport protocol in order to avoid the problems with TCP transport outlined above. This approach allows the DHT more freedom in which nodes it can contact, more control over the total load it places on the network, and better integration between the DHT's failure handling and packet retransmission.

DHash++ allows the option to use a specialized transport called the Striped Transport Protocol (STP). STP allows nodes to put and get data directly to other nodes, rather than routing the data through multiple overlay hops. STP does not maintain any per-destination state; instead, all of its decisions are based on aggregate measurements of recent network behavior, and on Vivaldi latency predictions. STP's core mechanism is a TCP-like congestion window controlling the number of concurrent outstanding RPCs.

While STP borrows many ideas from TCP, DHT data transfers differ in important ways from the unicast transfers that TCP is designed for. Fetching a large quantity of DHT data involves sending lookup and get requests to many different nodes, and receiving data fragments from many nodes. There is no steady "ACK clock" to pace new data, since each RPC has a different destination. The best congestion window size (the number of outstanding RPCs to maintain) is hard to define, because there may be no single delay and thus no single bandwidth-delay product. Quick recovery from lost packets via fast retransmit [41] may not be possible because RPC replies are not likely to arrive in order. Finally, averaging RPC round-trip times to generate time-out intervals may not work well because each RPC has a different destination.

The rest of this section describes the design of STP.

5.2.1 STP window control

Each DHash++ server controls all of its network activity with a single instance of STP. STP maintains a window of outstanding UDP RPCs: it only issues a new RPC when an outstanding RPC has completed. STP counts both DHT lookup and data movement RPCs in the window.

STP maintains a current window size w in a manner similar to that of TCP [21, 9]. When STP receives an RPC reply, it increases w by $1/w$; when an RPC times out, STP halves w.

STP actually keeps $3w$ RPCs in flight, rather than w. Using w would cause STP to transfer data significantly slower than a single TCP connection: lookup RPCs carry less data than a typical TCP packet, STP has nothing comparable to TCP's cumulative acknowledgments to mask lost replies, STP's retransmit timers are more conservative than TCP's, and STP has no mechanism analogous to TCP's fast retransmit. The value 3 was chosen empirically to cause STP's network use to match TCP's.

5.2.2 Retransmit timers

Lost packets have a large negative impact on DHash++ throughput because each block transfer is preceded by a multi-RPC lookup; even a modest packet loss rate may routinely stall the advancement of the window. Ideally STP would choose timeout intervals slightly larger than the true round trip time, in order to to waste the minimum amount of time. This approach would require a good RTT predictor. TCP predicts the RTT using long-term measurements of the average and standard deviation of per-packet RTT [21]. STP, in contrast, cannot count on sending repeated RPCs to the same destination to help it characterize the round-trip time. In order for STP to perform well in a large DHT, it must be able to predict the RTT before it sends even one packet to a given destination.

STP uses Vivaldi latency predictions to help it choose the retransmit time-out interval for each RPC. However, Vivaldi tends to under-predict network delays because it does not immediately account for current network queu-

ing delays or CPU processing time at each end. Since under-predicting the latency of an RPC is costly (a spurious loss detection causes a halving of the current window) STP adjusts the Vivaldi prediction before using it. STP characterizes the errors that Vivaldi makes by keeping a moving average of the difference between each successful RPC's round-trip time and the Vivaldi prediction. STP keeps this average over all RPCs, not per-destination. STP chooses an RPC's retransmission interval in milliseconds as follows:

$$RTO = v + 6 \times \alpha + 15 \quad (2)$$

where v is the Vivaldi-predicted round trip time to the destination and α is the average error. The weight on the α term was chosen by analyzing the distribution of RPC delays seen by a running node; the chosen timers produce less than 1 percent spurious retransmissions with approximately three times less over-prediction in the case of a loss than a conservative (1 second) timer. This formula assumes that Vivaldi's errors are normally distributed; adding a constant times the error corresponds to sampling a low percentile of the error distribution. The constant α plays a part similar to the measured RTT deviation in the TCP retransmit timer calculation.

The constant term in Equation 2 (15 ms) is necessary to avoid retransmissions to other virtual nodes on the same host; Vivaldi predicts small latencies to the local node, but under high load the observed delay is as much as 15 ms. This term prevents those retransmissions without adding significantly to over-prediction for distant nodes.

5.2.3 Retransmit policy

When an STP retransmit timer expires, STP notifies the application (DHash++) rather than re-sending the RPC. This gives DHash++ a chance to re-send the RPC to a different destination. DHash++ re-sends a lookup RPC to the finger that is next-closest in ID space, and re-sends a fragment fetch RPC to the successor that is next-closest in predicted latency. This policy helps to avoid wasting time sending RPCs to nodes that have crashed or have overloaded access links.

DHash++ uses a separate background stabilization process to decide whether nodes in the finger table or successor list have crashed; it sends periodic probe RPCs and decides a node is down only when it fails to respond to many probes in a row.

5.3 Performance comparison

This section presents measurements comparing the latency and throughput of the TCP transport implementation to the STP implementation when run on the RON test-bed. We used 26 RON nodes, located in the United States and Europe. Each physical RON node is located in a different machine room and ran 4 copies of DHash++. The average inter-node round-trip time is 75 ms, and the median is 72 ms (these reflect the multiple copies of DHash++ per host).

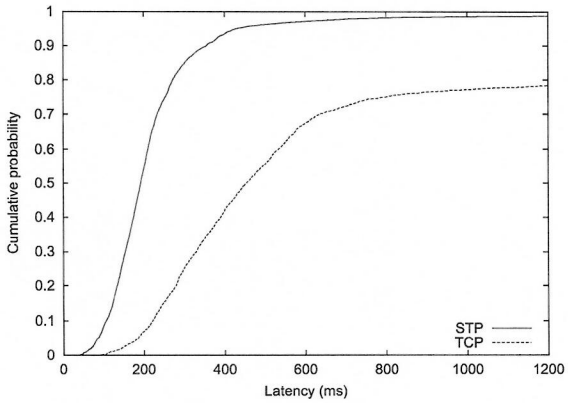

Figure 13: Distribution of individual 8192-byte fetch latencies on RON.

5.3.1 Fetch latency

Figure 13 shows the distribution of individual block fetch latencies on RON. The numbers are derived from an experiment in which each node in turn fetched a sequence of randomly chosen blocks; at any given time only one fetch was active in the DHT. The median fetch time was 192 ms with STP and 447 ms with TCP. The average number of hops required to complete a lookup was 3.

The STP latency consists of approximately 3 one-way latencies to take the lookup to the predecessor, plus one one-way latency to return the lookup reply to the originator. The parallel fetch of the closest seven fragments is limited by the latency to the farthest fragment, which has median latency (see Section 4.4). Thus the total expected time is roughly $4 \times 37.5 + 72 = 222$; the actual median latency of 192 ms is probably less due to proximity routing of the lookup.

The TCP latency consists of the same three one-way latencies to reach the predecessor, then a median round-trip-time for the predecessor to fetch the closest seven fragments, then the time required to send the 8 KB block over three TCP connections in turn. If the connection uses slow-start, the transfer takes 2.5 round trip times (there's no need to wait for the last ACK); if not, just half a round-trip time. A connection only uses slow-start if it has been idle for a second or more. The connection from the first hop back to the originator is typically not idle, because it has usually been used by a recent fetch in the experiment; the other connections are much more likely to use slow start. Thus the latency should range from 340 ms if there was no slow-start, to 600 ms if two of the hops used slow-

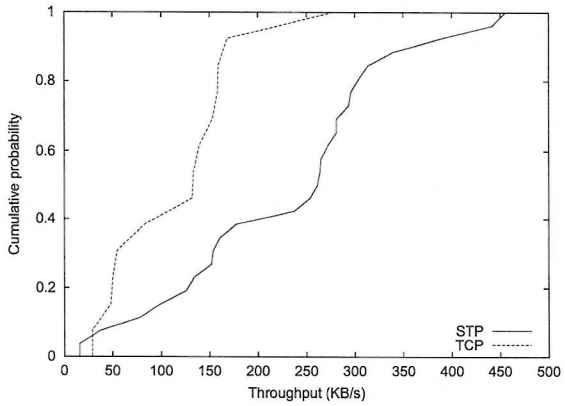

Figure 14: Distribution of average throughput obtained by different RON nodes during 4 megabyte transfers.

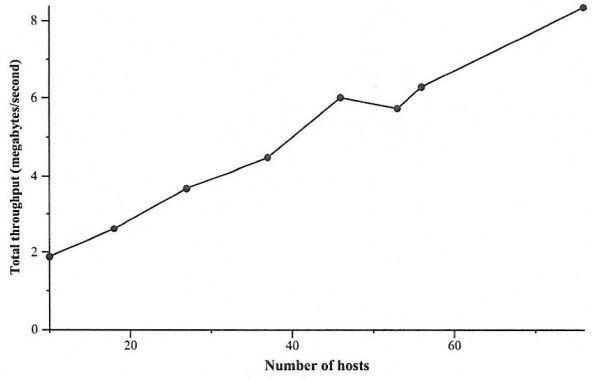

Figure 15: The effect of system size on total throughput obtainable. Each point represents an experiment with DHash++ running at x sites on the RON and PlanetLab test-beds. Each site reads 1000 8 KB blocks; the aggregate throughput of the system in steady state is reported. This throughput increases as additional capacity (in the form of additional sites) is added to the system.

start. The measured time of 447 ms falls in this range. This analysis neglects the transmission time of an 8 KB block (about 131 ms at 1 Mb/s).

5.3.2 Single-client fetch throughput

Figure 14 shows the distribution of fetch throughput achieved by different RON nodes when each fetches a long sequence of blocks from DHash++. The application maintains 64 one-block requests outstanding to its local DHash++ server, enough to avoid limiting the size of STP's congestion window.

Using TCP transport, the median node achieved a throughput of 133 KB/s. The minimum and maximum throughputs were 29 and 277 KB/s. Both the median throughput and the range of individual node throughputs are higher when using STP: the median was 261 KB/s, and throughputs ranged from 15 to 455 KB/s. The TCP transport has lower throughput because it sends each block back through each node on the recursive route, and thus is more likely than STP to send a block through a slow access link. About half of the three-hop routes pass through one of the RON sites with sub-one-megabit access links. STP sends coded fragments directly to the node originating the request, and thus each fragment encounters fewer slow links.

To characterize the effectiveness of STP in utilizing available resources we consider the expected throughput of a DHash++ system. Assuming an STP window large enough to keep all links busy, a node can fetch data at a rate equal to the slowest access link times the number of nodes, since the blocks are spread evenly over the nodes.

The slowest site access link in RON has a capacity of about 0.4 Mb/s. With 26 nodes one would expect $0.4 \times 26 = 10.4$ Mb/s or 1.3 MB/s total throughput for a fetching site not limited by its own access link. STP achieves less than half of this throughput at the fastest site. The reason appears to be that STP has difficulty maintaining a large window in the face of packet loss, which averages about 2 percent in these tests.

5.3.3 Scale

This section evaluates the ability of STP and DHash++ to take advantage of additional resources. As the number of nodes grows, more network capacity (in the form of additional access links) is added to the system. Figure 15 shows the total throughput for an N-node DHT when all N nodes simultaneously read a large number of blocks, as a function of N. The experiments were run on the combined PlanetLab and RON test-beds. The slowest access link was that of a node in Taiwan, which was able to send at 200 KB/s to sites in the US. The observed throughput corresponds to our throughput prediction: the total throughputs scales with the number of sites. The first data point consists of ten sites experiencing an aggregate of ten times the bandwidth available at the slowest site.

A similar experiment run using 150 machines but at 70 unique sites (many PlanetLab sites are home to more than one node) produces a peak throughput of 12.8 MB/s. As more machines and DHash++ nodes are added to each site, that site gains a proportionally greater share of that site's link bandwidth and the system's aggregate bandwidth increases.

6 Related work

The primary contribution of this paper is exploring a large set of design decisions for DHTs in the context

of a single, operational system. This exploration of design decisions emerged from an effort to understand a number of recent DHT-like systems with different designs, including OceanStore/Pond [22, 34, 43], CFS [13], Overnet [28, 32], PAST [6, 37, 38] FarSite [1], and Pastiche [11].

Rhea *et al.* did a black-box comparison of the implementations of several structured lookup systems (Chord, Pastry, Tapestry) and found that the use of proximity information reduced lookup latency, especially for lookups destined to nearby hosts [36].

Gummadi *et al.* studied the impact of routing geometry on resilience and proximity [16]. They found that flexibility in the routing geometry in general improved the ability of the system to find good neighbors. We extend their findings with additional analysis in simulation and with actual measurements to better understand the performance gains seen when using proximity.

A number of recent papers discuss design ideas related to networks with churn [5, 23, 35], some of which are used by DHash++. These ideas include integrated transport systems that quickly detect node failure and use alternate routes after timeouts.

7 Conclusions and future work

This paper has presented a series of design decisions faced by DHTs that store data, discussed the design options and how they interact, and compared a number of variant designs using simulations and measurements of an implementation running on PlanetLab and RON. The paper proposed techniques that taken together together reduce fetch latency by a factor of two and allow efficient bulk throughput.

The list of design decisions is not exhaustive, and future work will analyze a wider range of DHT designs and behavior such as the relationship between lookup robustness and performance, the latency and throughput of DHT writes, the handling of data movement required when nodes join and leave, data layout policies, the effects of block size, and the tradeoffs involved in use of constant-hop-count lookup protocols.

The simulator and DHash++ are publically available from `http://project-iris.net`.

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This paper is dedicated to the memory of Josh Cates.

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