Streaming Live Media over a Peer-to-Peer Network

Hrishikesh Deshpande
CSD, Stanford University
hrishi@cs.stanford.edu

Mayank Bawa
CSD, Stanford University
bawa@cs.stanford.edu

Hector Garcia-Molina
CSD, Stanford University
hector@db.stanford.edu

Abstract

The high bandwidth required by live streaming video greatly limits the number of clients that can be served by a source. In this work, we discuss and evaluate an architecture, called *SpreadIt*, for streaming live media over a network of clients, using the resources of the clients themselves. Using SpreadIt, we can distribute bandwidth requirements over the network. The key challenge is to allow an application level multicast tree to be easily maintained over a network of transient peers, while ensuring that quality of service does not degrade. We propose a basic peering infrastructure layer for streaming applications, which uses a redirect primitive to meet the challenge successfully. Through empirical and simulation studies, we show that SpreadIt provides a good quality of service, which degrades gracefully with increasing number of clients. Perhaps more significantly, existing applications can be made to work with SpreadIt, without any change to their code base.

1 Introduction

Live streaming media will form a significant fraction of the internet traffic in the near future. Recent trade reports indicate that if the current acceptance rate among end-users persists, streaming media could overtake television with respect to the size of the client base [Dor01].

Since video streams are fundamentally high bandwidth applications, even a small number of clients receiving the stream are often sufficient to saturate the bandwidth at the source. For example, a T3/DS3 connection has a capacity of 45 Mbps, while a stream with 30 fps, at 320x240 pixels can have a rate of 1 Mbps. Under such conditions, only 45 clients can be provided a maximum resolution video stream. IP Multicast, if and when deployed, will eliminate the source bandwidth bottle-neck by feeding many clients from a single stream. Until then, web sites, working under a limited bandwidth resource, have to either degrade the quality of service (QoS), or turn away a large number of clients. We present an incident below, on site outage due to unexpected flash crowds at a live event streaming site. Indeed, it is only one of a number of incidents that have been reported recently.

EXAMPLE 1.1 *In April 2001, Doordarshan Online (at http://www.dd.now.com) used Akamai's content distribution network to web-cast India-Australia cricket matches. The company (dd.now.com) had provisioned a certain bandwidth from Akamai with its average number of clients in mind. As the matches approached a nail-biting finish, the number of clients requesting the feed increased to exceed the provisioned bandwidth. Rather than turn clients away, the*

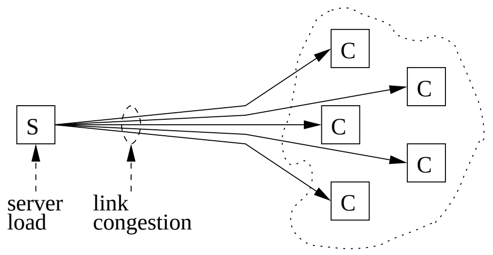


Figure 1: Current solutions cause congestion at server and gateways

web-site cut off its higher resolution feed (designed for T1 clients), and provided only a lower resolution feed (designed for DSL clients). In the final hour of the match, several clients on the West Coast were refused a feed, leading to disrepute of the site and frustration among end-users[1]. □

In the example above, the problem was exacerbated because a large number of clients were concentrated in a small geographic area (Stanford University). As can be seen in Figure 1, the path from the server to Stanford University got congested as each data packet was replicated and sent over the link as many times as the number of clients. As a consequence, the QoS perceived by the end-user degraded.

Current trends indicate that such problems will aggravate in the near future. Increasingly, people are moving away from modems to DSL, enabling users to request multimedia feeds from various sources. It is reasonable to expect that as the edge bandwidth improves, the size of flash crowds will increase, corresponding to taller spikes in traffic. As the potential for demand increases, more events will be web-cast live by different companies to feed an increasing client base.

Designing a content distribution system which can scale with the number of clients is a challenging task. At the same time, such a system needs to be cost effective. The two requirements effectively rule out any attempt at increasing the resources of the source server. Distributed systems which use replication (e.g., Akamai) to spread out bandwidth form a viable alternative, but even such systems have a hard upper bound on the number of clients that can be supported for a given price, and a high cost associated with provisioning required bandwidth.

Single source multicast is a solution because it uses a single stream to feed all the clients. Multicast solutions have been attempted at various layers of the network stack. Link-layer multicast is primarily constrained by the fact that groups are statically set up by the network provider, and can be changed only by a reconfiguration of the network [Mil98]. IP multicast is more flexible, but requires support at the routers. The complexity costs of group management, distributed multicast address allocation, and support for network management have limited the commercial deployment of IP multicast [DLL$^+$00]. Application layer multicast has been more popular, and been deployed successfully by infrastructure companies [Fas, Dig]. Reliable servers are installed across the Internet, to act as routers by splitting and forwarding the

[1]Two of the frustrated cricket fans are authors of this paper; the incident motivates the work reported here.

stream to clients in the same intranet. Here, the complexity and cost arises from deploying and maintaining the servers.

In such a context, it seems desirable to turn the problem on its head, and use the bandwidth resources of the clients themselves to multicast the stream. Peer to peer networks hold the promise of such a paradigm. However, building a stream distribution system from thin and unpredictable nodes, and yet providing a reasonable QoS, is a challenging problem.

In this paper, we propose a solution, called *SpreadIt* which builds an application level multicast tree over the set of clients. Each client node needs to be enabled with a basic *peering* layer between the application and transport layers. A node can function as a server and a client simultaneously. As a client, it receives the stream from some node in the network. As a server, it forwards the stream to other nodes in the network. The application (RealPlayer, Windows Media Player, etc.) gets the stream from the peering layer on their local machines. The peering layers at different nodes coordinate among themselves to establish and maintain a multicast tree. Since the clients are autonomous, subscribes and unsubscribes are unpredictable. The key challenge is in masking the transience of nodes while ensuring a good QoS. The peering layer is responsible for hiding changes in the multicast topology from the applications above.

As we show later, the SpreadIt architecture can support (to the order of) thousands of clients, and allows relatively cheap deployment. It enables a good QoS, which degrades gracefully with increasing number of clients. Importantly, existing applications do not need to be changed to use the peering layer, and are thus enabled for such a stream distribution. On the negative side, SpreadIt does not provide rigid guarantees of reliability and availability as provided by high end commercial vendors providing dedicated multicast servers. Moreover, clients need to download, and execute the peering layer to enable the distribution network to be built. We have implemented SpreadIt which is available for download at `http://www-db.stanford.edu/peers`.

The organization of the rest of the paper is as follows. In Section 2, we give a detailed model, and definitions for our peer-to-peer network. In Section 3, we present our stream distribution architecture, and indicate the issues to be resolved. In Section 4, we describe the primitives and mechanisms used to enable the architecture. In Section 5, we discuss the various policies evaluated for maintaining the topology of the multicast tree. In Section 6, we evaluate the various policies on the proposed architecture with respect to the QoS enabled. In Section 7, we discuss related work. In particular, we position our work in the context of application level multicast solutions. We conclude with a summary in Section 8. In an extended Technical Report [DBGM01], we show how to use SpreadIt with existing implementations of the end applications.

2 Model

In this section, we give precise definitions for our peer-to-peer network model.

A stream is a time ordered sequence of packets. A stream is logically composed of two channels: *data* and *control*. The stream has two end-points: a server which feeds the data to a client, and a client which receives the data from a server. The data channel comprises of the actual payload packets. The control channel is used by a client and its server to coordinate. Examples of coordination are connection setup, teardown, play, stop, etc. The stream is served over the Internet, using standard streaming protocols (RTP for data and RTSP for control channels). We note that usually data is sent over RTP/UDP [SCFJ96], while control messages are sent over RTSP/TCP [SRL98].

Let s denote the source of a live stream. We assume that the source, s, remains up for the duration of the stream. We also assume that each live stream has a unique URI (Uniform Resource Identifier), known to all potential clients. The URI has information about s embedded in it.

A client node wishing to receive the stream sends a *subscribe* request to the source. The client can be directed by the source to a server which can support the client. When the client wants to stop receiving the stream, it sends an *unsubscribe* request to its server. At any instant, a set of nodes (clients), $\aleph$, are receiving the stream. The set $\aleph$ changes over time, as nodes subscribe or unsubscribe.

An instance of the data and control channels, taken together, between a server and a client constitutes a *data-transfer session*. Such a session is distinguished from the *application session* which begins when a client subscribes to a stream from a source, and ends when the client unsubscribes. The application session can subsume several data-transfer sessions as the client changes servers from which it receives feed for a given stream.

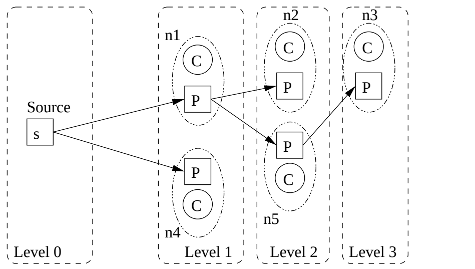
A *live* stream has the important property of being *history-agnostic*: the client is only interested in the stream from the instant of its subscription onwards.

Each node, n, has a unique identifier (IP address) which remains fixed and unique over its application session. However, each node is autonomous, and can subscribe or unsubscribe from a stream at will, even without notification. Each node is connected to the Internet with a certain bandwidth capability. An overlay network formed on a set of autonomous nodes, in which a node can simultaneously serve some nodes, and be served from another node, is a *peer-to-peer* network. A node which is unable to serve an incoming client c with an acceptable QoS is said to be *saturated* with respect to c, otherwise it is *unsaturated.* A node which expects the stream, but is not receiving it is said to be in a *transient* state. The time taken by the node to recover from a transient state is called *transience* time. For example, a client which had sent a subscribe request to s, but has not received the stream yet is in a transient state.

The goal is to design an efficient architecture for feeding a set of nodes, $\aleph$, with a particular live stream sourced from s while providing a high quality of service. Typically, QoS is characterized by packet loss, packet delay, time to first packet (time elapsed between a subscribe request send and the start of stream), and jitter. Jitter is effectively eliminated by a huge client side buffer [SJ95]. Hence, we shall not worry about jitter in the rest of the paper.

3 The SpreadIt Architecture

The architecture we propose performs multicast at the application-layer. A multicast tree (see Figure 2) rooted at s, is built over the set of nodes, $\aleph$. Nodes are organized into different levels. For each node, n, at level $l+1, l=0,1,2...$, there is a node, called its *parent*, p, at level l; n is called a *child* of p. All nodes in the sub-tree rooted at p are called its *descendants.*



N = {n1, n2, n3, n4, n5}, P: Peering Layer, C: Application Client

Figure 2: An application level multicast tree built on the peers

The multicast tree is built incrementally, as nodes subscribe to the stream. A data-transfer session is established between a node and each of its children. The source, s, feeds the stream to its children, which are the level 1 nodes. Each node, n, at level l forwards the stream to all its children (if any) at level $l+1$. Thus, each node acts simultaneously as a client (to its parent) and a server (to its children). Effectively, each node acts as a router, forwarding packets to its children, and thus, the nodes multicast the original stream.

An application level multicast helps to balance out the server bandwidth load. To illustrate, assume each node can support d clients, and that the nodes have arranged themselves to form a l level complete tree. Thus, $|\aleph| = O(d^l)$. If the server were feeding all clients by itself, it would need to send $|\aleph|$ packets per time unit. Under the SpreadIt architecture, it has to feed only d nodes. As a result, congestion is removed from the network near the server, and the bandwidth requirements are spread over the network of clients.

However, for the above benefits to be realized, the following issues need to be resolved. We shall discuss the solutions in Sections 4 and 5.

1. *Discovering a Server*: The multicast tree is built incrementally as nodes subscribe to the stream. New nodes are supported by nodes already subscribed to the stream. There needs to be a mechanism to enable the new nodes joining the multicast to identify an existing node in the tree, which can act as its parent.

2. *Managing Transience*: Nodes can unsubscribe from the multicast network at will. If a node opts out, all its descendants are left stranded. It is essential that such disconnections be repaired for the descendants to continue receiving the feed, and that too, in a duration which has minimal effect on the QoS perceived by its descendants.

3. *Managing Load*: The multicast tree is built over nodes whose primary intent is to act as individual clients. The resources of the nodes should be judiciously used while constructing the multicast tree, so as not to overload any node. Most of the nodes today have significant processing power and memory resources. The constraining resource at a node is predominantly bandwidth capacity. Each node is aware of its own capacity, and can, for a given stream bit-rate, fix for itself a maximum limit on the number of children it will support.

Note that, using an application-layer multicast solution results in increased packet loss and delay at a node. A node at level i receives data along a path of length i from the source. At each hop, packets propagate across the network, work their way up through the network stack to the application layer, and then back down to be transmitted along the next hop. The delay of a packet perceived at a node is the cumulative of delays at each intermediate node. In particular, the data channel of the stream typically uses UDP, which does not provide guarantees on reliable delivery of packets. A level i node receives data from the source over a path of i hops. So the packet losses and delay accumulate across each hop. Our experiments (Section 6) suggest that the effects of packet loss and delay are limited, and a good QoS can be enabled using SpreadIt.

4 Peering Layer: Primitives and Mechanisms

4.1 The Peering Layer

In this subsection, we describe the peering layer needed by the SpreadIt, at each client. for such a layer, and then provide the specifications for the layer.

The multicast tree is built over the client nodes themselves. However, the nodes are autonomous and unpredictable. If a node at an intermediate level of the tree unsubscribes from the stream, the stream is cut off from all its descendants. Clearly, the disconnected sub-trees need to be grafted onto the main source rooted tree.

However, the tree itself is composed of data-transfer sessions established between a parent and a child. The data-transfer sessions are built on unicast transport protocols (UDP for data, TCP for control). In current implementations, a tight coupling exists between the transport layer, and the application layer. If during a unicast data-transfer, the server fails, the application using the data at the client crashes. The application is unable to survive even if there exist alternate available sources providing the same data feed. Thus, we note that application session semantics are *not preserved*, due to the *inability* of the lower layers to mask the failure of a data-transfer session.

In our application level multicast, each node of the tree receives the same data (barring packet losses). The parent-child connection for the feed is, at best, incidental. The multicast tree topology exists only to ensure that data reaches all the participating nodes. Ideally, the application session at a node should survive failure of the data-transfer session, and persist while a new parent is located, and the failed data-transfer session is restarted.

The key to tackle problems arising from transient sources is based on the following simple insight: the tight coupling between application layer and data-transfer layer can be broken by a level of indirection.

We introduce a basic peer-to-peer infrastructure layer between the application and the transport layers for streaming. All communication between the application and the transport layers passes through the peering layer, as shown in Figure 3.

Data transfer sessions are established between the peering layers at the two nodes. The end-points are identified by the tuple ⟨ *Server IP address, Server port, Client IP address, Client port, Stream URI* ⟩. The first four components identify the transport end-points involved in the session. The last component serves to identify the specific stream data is intended for.

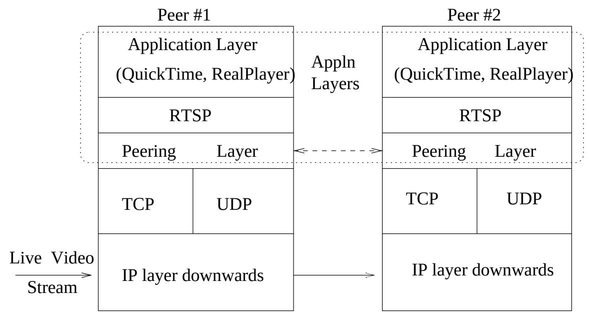


Figure 3: A layered architecture of a peer

The peering layer allows the application layer above to specify the stream to obtain, through a "get-stream" interface. Given a stream URI, the peering layer locates a server which can provide the stream, negotiates with the server for opening a data-transfer session, and establishes a transport session to the server. It interacts with the peering layer at the server to detect, or report a data-transfer session termination. In the event of a data-transfer session termination, it cleans up the transport layer session, locates a new server for data feed, and restarts the flow of data. If an alternate server cannot be found, an error is flagged to the application above. The application survives these changes, and is *unaware* and, indeed, *unconcerned*, of shifts in the underlying multicast topology, as it should be. Furthermore, the code of existing applications does not need to be changed to enable the use of our peering layer (see [DBGM01]). We also discuss the design philosophy for the peering layer, and discuss its end-point addressing and guarantees in the extended version of the paper.

The guarantees it provides reflect those of the transport layer protocol used in the data-transfer session (for which it has to maintain state). In addition, it guarantees that the data feed to the application above will be maintained as long as an unsaturated server can be found.

4.2 The Redirect Primitive

The peering layer supports a redirect primitive used to effect changes in the topology. The primitive is simple and light-weight. It hides a lot of implementation details, and is a good building block for tree-maintaining algorithms. The peering layer uses redirects as hints, to enable discovery of an unsaturated node in the network.

As shown in Figure 4, a redirect message is sent by a peer, p, to another peer, c, which is either opening a data-transfer session with p, or has a session already open. The message specifies a target peer, t. On receipt of the redirect message, c closes its data-transfer session with p, and tries to establish a data-transfer session with t, for the same stream URI. Note that such redirect messages are produced and used at the peering layer, and the application above is unaware of such messages. Thus, it is the redirect primitive which allows the application session to persist despite changes in the data-transfer sessions.

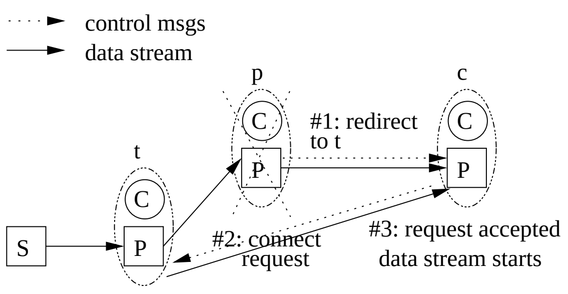


Figure 4: An example of the redirect primitive in use

4.3 Discovering an Unsaturated Server

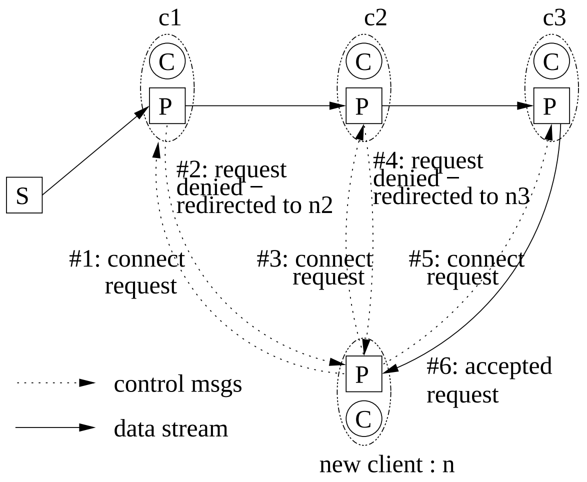


Figure 5: Adding a new node to a multicast tree

A new node, n, seeking the live stream needs to be able to discover an unsaturated node in the peer-to-peer network, which can act as its parent. Such a resource discovery mechanism is built using the redirect primitive.

The process of addition of a new node is shown in Figure 5. To start the process, n contacts the source, s, of the stream at the universally known URI. When s receives a data-transfer session setup request, it checks to see if it is itself unsaturated. If s is unsaturated, it accepts n as its child and starts feeding it the stream by establishing the data-transfer session. If s is saturated, it responds to n with a redirect message with one of its immediate children c as the target peer. Upon receiving the redirect, n attempts to setup a data-transfer session with c. The process continues iteratively, and n gets bumped down the multicast tree until its is accommodated. If n is unable to find an unsaturated node within some specified number of tries, the peering layer flags a resource unavailable error to the upper application-layer.

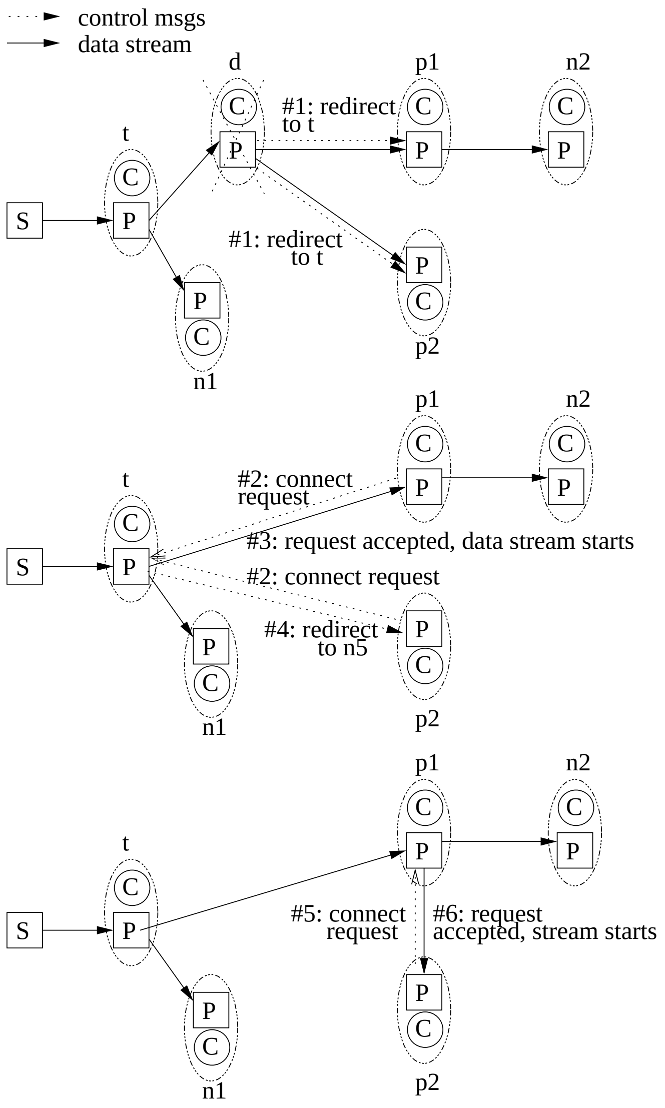


Figure 6: Delete of an intermediate node from a multicast tree

4.4 Managing Deletion of Nodes

When one of the nodes, d, in a multicast tree unsubscribes from the stream, the descendants of d become transient. To recover from such an event, the descendants have to be grafted back into the source-rooted tree after d has been removed.

The simplest case is when d was acting just as a client, and was not serving any other node in the network. Then, d sends an unsubscribe request to its parent. The parent frees up resources dedicated to d at its side. No further changes in the tree are needed.

The process of deleting an intermediate node is shown in Figure 6. If d was the direct parent of a set of nodes C, it also sends a redirect message to all the nodes in C specifying a target t. The value of t might be the parent of d, or the source s. The nodes in C then start the process of finding an unsaturated server by contacting t, as discussed above.

4.5 Handling Failures of Nodes

Intermediate nodes might fail, without being able to inform their parent, or send redirect messages to their children. The network needs to first detect such a failure, and then recover from it. The peering layer at a node uses a heart-beat mechanism to send alive messages to its parent, and children. If a peer detects that a child has skipped a specified number of heart-beats, it deems the child as dead, and cleans up the data-transfer session at its end. If a peer detects that the parent has skipped a specified number of heart-beats, it deems the parent as dead. It then recovers from its transience state by sending a redirect message to itself specifying the source, s, as the target. The process of discovering an unsaturated server in the network, as mentioned in Section 4.3, is then started.

5 Policies for Topology Maintenance

Observe that the tree topology is defined by specifying the policy for choosing a target peer in a redirect message. Such a policy will have implications on keeping the tree balanced. The time taken to discover an unsaturated server depends on the policy, which in turn affects the the time to first packet of a new node, and the transience time of nodes. The policy also determines the information about the network that needs to be maintained at each peer.

A topology maintenance policy must specify the target peer to contact under addition, and deletion (or failure) of nodes. Such a policy can then be implemented using the mechanisms we discussed in Section 4. The tree can be optimized with respect to a variety of cost functions (available resource, bandwidth, delay, packet loss). For a cost function, a host of policies are feasible, ranging from light-weight to a completely centralized heavy-state policy.

In this section, we shall discuss some of the policies, based on simple cost functions, that are appealing because of their light-weight and distributed nature. In particular, we assume that each node knows only the source s, and its local topology: its parent, and the set of children it supports.

5.1 Addition of a peer (or Add Policies)

A node which is unsaturated always accepts a data-transfer session setup request. However, a saturated node, Y, needs to forward the requesting client X to another peer in the network which is also getting the stream feed. Since a peer only knows its local topology, Y can only forward X to one of Y's immediate children, or its parent. Some of the options in choosing such a target are the following:

1. *Random* : Y chooses one of its children at random as the target t, and redirects X to t. Such a policy requires minimal state at Y. On an average, the tree is expected to be balanced.

2. *Round-Robin (RR)* : Y maintains a list of its children. Y forwards X to the child, t, at the head of the list. The child, t, is then moved from the head to the end of the list. Such a policy requires some state maintenance, but is expected to keep the tree balanced.

3. *Smart-Placement (SP)* : X sends traceroute information along-with its request. Y maintains the network locations of its children. Y redirects X to a child that is closest to X [FJJ$^+$01]. Such a policy helps in creating trees taking network proximity into account. Packet losses and delays are expected to be minimized, thereby improving the QoS metric.

4. *Knock-Downs* : X sends its traceroute information along-with its request. Y maintains the network locations of its children. If X is closer to Y than any of Y's children, Y redirects its child which is the farthest, and accepts X as a child. Otherwise, it redirects X to a child that is closest to X. Such a policy helps in creating trees taking network proximity into account. Packet losses and delays are minimized.

5. *Smart-Bandwidth* : X sends its bandwidth capacity along-with its request to Y. If the bandwidth capacity of Y is greater than or equal to that of X, act according to one of the above policies. Otherwise, Y closes its data-transfer session with its parent, t. It then redirects X to t, and redirects itself and all its immediate children to X.

5.2 Deletion of a peer (or Delete Policies)

When a node X wants to unsubscribe, it needs to forward a suitable valid target t to its descendants. Each node is definitely aware of two nodes in the network which get a stream feed independent of itself: its parent, and the source. Thus, there are at least two candidate values for t. When X is unsubscribed, all of its descendants become transient. Each transient descendant can try to recover by contacting t. Alternately, only the children C of X attempt to recover by contacting t. The rest of the descendants of X, are also the descendants of C. Thus, these nodes automatically recover when the feed to C is restored. Hence, we have the following policies.

1. *Grandfather-All (GFA)* : X chooses its parent (which is the grandfather of C) as t. X sends a redirect message to all its children to contact t. Each child of X in turn sends

a redirect message to its children with t as target, and so on. Thus, all the descendants of X try to recover by contacting t. The advantage of such a policy is that the height of the affected portion of the tree can be reduced.

2. *Root-All (RTA)* : X chooses the source as t. As in GFA, all the descendants of X attempt to recover by contacting t. The advantage is that the affected nodes move closer to the source, reducing the height of the tree.

3. *Grandfather (GF)* : X chooses its parent (which is the grandfather of C) as t. Only nodes in C attempt to recover by contacting t. The rest of the descendants rely on C to restore their feed. The advantage of such a policy is that the effects of failures are localized. Other nodes in the tree are not affected. Moreover, it avoids the explosion of requests to t as in GFA.

4. *Root (RT)* : X chooses the source as t. As in GF, only nodes in C attempt to recover by contacting t. The advantage of such a policy is that the sub-trees rooted in C could be accommodated near the source, causing fewer packet losses, and delays for the affected nodes. Moreover, it also avoids the explosion of requests to t.

Note that the policy to recover from failure is similar to delete, once a failure of the parent is detected. However, in this case, the identity of the parent of the failed node is not known to the descendants of the node. Hence, only the Root and Root-All policies are relevant here.

6 Performance Evaluation

In this section, we study the performance of SpreadIt. We implemented SpreadIt, and obtained values for basic parameters through empirical studies. The values were then used in simulations to evaluate the various policies for the QoS metrics.

6.1 Experimental Testbed

We used Apple's open source Darwin Streaming Server at the source to simulate a live broadcast by looping over a video clip. Apple's QuickTime was used as the application level client at the peer nodes. The peering layer was implemented in Python. The peering layer used the redirect mechanism to discover unsaturated nodes during addition, deletion, and failures. The add policy was set to Random, and the delete policy to Root (RT). Failures of nodes were detected by the absence of heartbeats. The implementation details are discussed in Appendix A.

We performed experiments on Sun Ultra 60 computers with 450MHz UltraSPARC-II processors, 256MB RAM, 1GB swap disk and 6.3GB disk running Solaris 7. The machines were connected with a 100Mbps ethernet. The video clip was 37.12s long, and 834,352B in size. The video resolution was 240x180, and the sound sampling rate was 11 KHz mono. The experiments were designed to measure the following parameters at a peer.

1. *RTP/UDP packet delays*, defined as the difference between the time a packet was sent by the source, and the time the same packet was received by a client. Note that packet

delay is caused by network latencies as well as processing time in a node's network stack and peering layer. Because of clock skew in machines, it is not correct to use timestamps from different machines. Instead, delay measurements were made by connecting clients in a loop so that the packet comes back to a different process on the same node which had initially sent it out. Thus, the timestamping is done by the same node which acts as the server, and the final client.

2. *Time to first packet*, defined as the time elapsed from the instant a new node sends a subscribe request to the source, to the instant the node receives the first stream packet. Both the times are with respect to the same node, and hence can be directly measured.

3. *Time to discover parent* : defined as the interval between the times a new node sends a subscribe request to the source, and to an unsaturated node. Thus, the time to discover parent measures the time spent in tree-transience management by a node. Both the time are wrt the same node, and hence can be directly measured.

4. *RTP/UDP packet drops* : We note that the encoding of a video stream has sufficient redundancy, such that a few isolated packet losses have almost no perceptible effect to human viewers. Thus, consecutive packet drops, called lost-blocks, are more important. The width of such lost-blocks, and the number of times they occur influences the QoS. RTP packets have a 16-bit field to store sequence numbers. The sequence numbers of all packets sent or received at a node were logged. The logs made at a client were compared against the log at the source. The difference in the logs indicates the dropped packets.

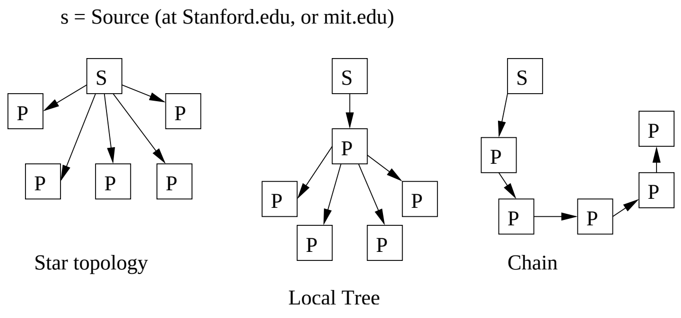


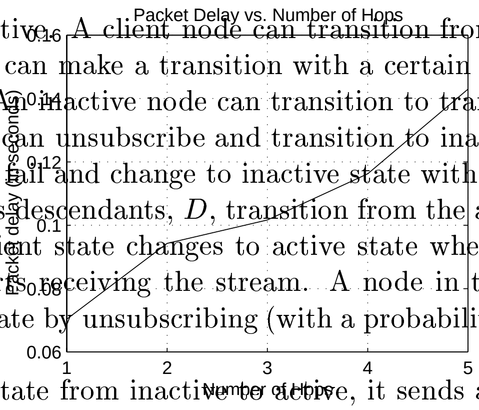
Figure 7: Topologies used in the experiments

In one set of experiments, the source was placed in the same intranet (in Stanford) as all the clients (peers). In the other set, the source was placed across the internet (at MIT) while all the clients were in the same intranet (in Stanford). Measurements were also made for different peer topologies as shown in Figure 7. A *Star* topology models the conventional architecture, in which all clients connect directly to a single source. We assume that in a Star architecture, the source has a sufficiently high bandwidth to support *all* the clients. Intuitively, Star models the best solution that does not use the aggregation provided by multicast. The *Local Tree* allowed us to study the effects of network proximity, while the *Chain* showed us the effects of increasing number of hops of a peer from the source.

Data was collected over five runs for each setting, and then averaged. Upto six clients were used in different experiments. Note that experiments results will be biased toward peers having a high bandwidth capacity. Hence, the reader should not interpret the results of this section as absolute predictions, but rather as illustrations of performance trends.

6.2 Results from Empirical Tests

1. *Packet Delays* increase linearly with the number of hops between the source and client. The average packet delay observed for a Chain topology over a 1 minute run, when the source and the clients were all within Stanford is shown in Figure 8. Note that the order of magnitude of the time delays, even at five hops, is about $0.1s$, while RealPlayer clients buffer the stream (at the application level) for upto $30s$. For the end-user, the delays due to buffering are 2 orders of magnitude greater than those due to increased hops. The linear growth implies that the height of the tree can be as high as 15, before hop delays become 1% of buffering delays. Thus, the number of clients supported can be in thousands (a tree has average height of 10, each node has 3 children, number of clients can be $(3^{11}-1)/2 = 88572)$ before delays become a dominating factor.



The source is always active. A client node can transition from one state to the other at each time step. Each node can make a transition with a certain probability, independent of the states of other nodes. An inactive node can transition to transient with a probability of p_{add}. If a node is active, it can unsubscribe and transition to inactive state with probability p_{unsub}. An active node can fail and change to inactive state with probability p_{fail}. If a node unsubscribes, or fails, all its descendants, D, transition from the active state to the transient state. A node in the transient state changes to active state when it gets assimilated in the source-rooted tree and starts receiving the stream. A node in the transient state can also transition to the inactive state by unsubscribing (with a probability of p_{unsub}) or failing (with a probability of p_{fail}).

When a node changes state from inactive to active, it sends a subscribe request to s (in the form of a connect request message). The peering system then accomodates the node into the source-rooted tree as specified by the *add policy*. A node which unsubscribes from the stream, sends a redirect message to its parent, and each of its children as specified by the *delete policy* in the next time step. A node which fails, just changes its own state to inactive. The parent of such a node, and each of its children detect the failure when the next heart-beat is missed. Each of the children (or descendants) then recover as specified by the *delete policy*, starting the recovery from the time instant the failure was detected.

To generalize the above model to clients spread across the internet, we extend the single intranet model by adding the following characteristics. Each node, and the source s, can belong to one of I intranets with an equal probability. A stream packet can be lost on a hop between two nodes in the same intranet with a probability of p_{li}, and across the internet with a probability of p_{lo}. Each node can support a maximum of c_{mi} clients in the same intranet, and a maximum of c_{mo} clients across an internet. The rest of the parameters, and modeling remains unchanged from the single intranet model.

The values for some parameters were measured from empirical runs, while others are "ed-

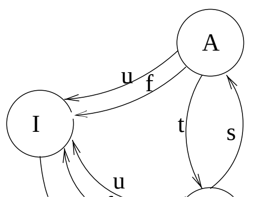


Figure 9: State transition diagram of a node

Table 2: Parameters used and their typical values.

Parameter(Symbol) for Intranet	Value
Packet Rate (t_s)	30 pkts/sec
Stream Duration (t_{len})	1 hour
Number of Clients (N)	1000
Maximum Children (c_{max})	10
Heart-Beat Interval (t_{hb})	1 sec
Connect Interval (t_c)	1/10 sec
Hand-Shake Interval (t_{hs})	1/2 sec
Start-Stream Interval (t_{hs})	1/30 sec
Prob. of Packet Loss (p_{loss})	1 in 100 pkts
Prob. of Addition (p_{add})	Once in 5 mins
Prob. of Unsubscribe (p_{unsub})	Once in 30 mins
Prob. of Failure (p_{fail})	Once in 50 mins
Number of Intranets (I)	10
Maximum Intranet Children (c_{mi})	10
Maximum Internet Children (c_{mo})	5
Pkt Loss Prob. over Intranet (p_{li})	1 in 100 pkts
Pkt Loss Prob. over Internet (p_{lo})	1 in 50 pkts

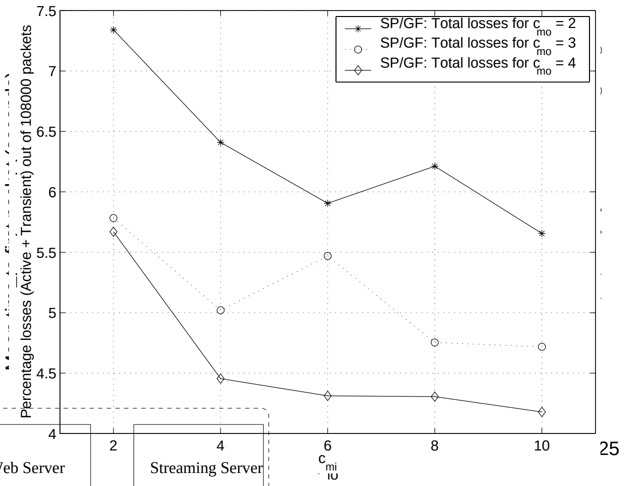
ucated guesses". The parameters and their values are listed in Table 2 for a ready reference.

6.4 Simulation Results and Inferences

We simulated Random, RR and SP as the add policies; GF, GFA, RT, and RTA as the delete policies; and compared the results for the QoS metrics. We also simulated the performance of a traditional solution, modeled as a Star topology. At the start of the simulation, only the source was active, and all the clients were inactive. Unless mentioned, the parameters have default values from Table 2. The stream was simulated for an hour, which is the duration of some sporting events. Statistics for each node were collected for a run, and then averaged over the number of nodes.

We simulated each combination of the add and delete policy for the multiple intranets model. The average number of lost packets observed by a node are shown in Figure 10. Star performs the best on the packet loss metric. However, note that in Star, the source is supporting 1000 clients directly, and thus has enough bandwidth to support 1000 replicated streams. For a much reduced workload (only 10 clients per node) SP/GF and SP/RT suffer only 2.5× more packet losses in the active state than Star. The increase factor is roughly equal to the average height of a node in the tree formed for the policy, and can be explained by the cumulative packet-loss effect. Thus, the increase in QoS may not be worth the high price of Star, under our assumptions.

The packet losses in the transient state in Star are nearly 4× more than SP/RT, even though a node enters a transient state more times in the SpreadIt architecture. Losses in the transient state are more important than those in the active state because the former occur in blocks, which can lead to stalls as perceived by the end-user. Since transience in Star corresponds to time to first packet (the only transience observed by a node is while waiting for the stream to begin), Star performs the worst among all policies on the metric. The



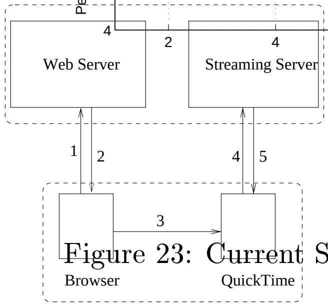


Figure 23: Current Streaming Site Architecture

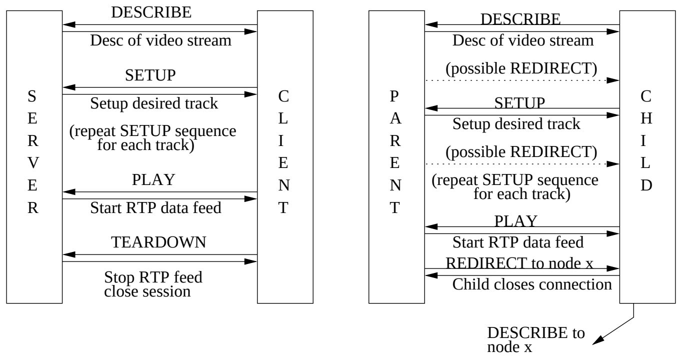


Figure 24: RTSP handshake between client and server

and all interactions between the streaming server, and the Media Player should be through the peering layer.

However, such a requirement can be worked around with the following trick. The site hosting the streaming server also hosts a peering enabled client, p. p receives a stream from the streaming server directly, and serves as a pseudo-source for the rest of the peering enabled clients. The web-server redirects all peering enabled clients to p, instead of the streaming server s.

At a peering enabled client, the QuickTime client is made to go through the peering application using the following scheme. The QuickTime client is provided the following RTSP url by the browser:
rtsp://localhost:cPort/peer?rtsp://streamingServer:sPort/broadcast, where *localhost:cPort* is the machine:port on which the peering layer is being run. The query string indicates the ultimate source of the stream. Once this is done, the QuickTime client communicates with the peering layer as if the peering layer were the source of the stream.

Now, the peering layer intercepts both the data (RTP) and control (RTSP) channels of the media stream, and passes messages from the client to the server, and vice-versa. The peering layers also interpret the peering messages to create and maintain the multicast tree by using the redirect primitive described in Section 4.2.

We have written scripts in Python to implement the peering layer, which can be downloaded from `http://www-db.stanford.edu/peers`. The current system has been tested on Solaris, Linux and Windows for Apple's QuickTime client. Unfortunately, each video player implements its own variant of RTSP, and so the current implementation is not compatible with Real systems.

Thus, we can wrap around both the streaming server, and the application client, and need not change either of them. However, it would be more efficient to incorporate the peering features in subsequent versions.