3.6 Principles of Congestion Control

In the previous sections, we examined both the general principles and specific TCP mechanisms used to provide for a reliable data transfer service in the face of packet loss. We mentioned earlier that, in practice, such loss typically results from the overflowing of router buffers as the network becomes congested. Packet retransmission thus treats a symptom of network congestion (the loss of a specific transport-layer segment) but does not treat the cause of network congestion—too many sources attempting to send data at too high a rate. To treat the cause of network congestion, mechanisms are needed to throttle senders in the face of network congestion.

In this section, we consider the problem of congestion control in a general context, seeking to understand why congestion is a bad thing, how network congestion is manifested in the performance received by upper-layer applications, and various approaches that can be taken to avoid, or react to, network congestion. This more general study of congestion control is appropriate since, as with reliable data transfer, it is high on our "top-ten" list of fundamentally important problems in networking. We conclude this section with a discussion of congestion control in the **available bit-rate (ABR)** service in **asynchronous transfer mode (ATM)** networks. The following section contains a detailed study of TCP's congestion-control algorithm.

3.6.1 The Causes and the Costs of Congestion

Let's begin our general study of congestion control by examining three increasingly complex scenarios in which congestion occurs. In each case, we'll look at why congestion occurs in the first place and at the cost of congestion (in terms of resources not fully utilized and poor performance received by the end systems). We'll not (yet) focus on how to react to, or avoid, congestion but rather focus on the simpler issue of understanding what happens as hosts increase their transmission rate and the network becomes congested.

Scenario 1: Two Senders, a Router with Infinite Buffers

We begin by considering perhaps the simplest congestion scenario possible: Two hosts (A and B) each have a connection that shares a single hop between source and destination, as shown in Figure 3.43.

Let's assume that the application in Host A is sending data into the connection (for example, passing data to the transport-level protocol via a socket) at an average rate of λ_{in} bytes/sec. These data are original in the sense that each unit of data is sent into the socket only once. The underlying transport-level protocol is a

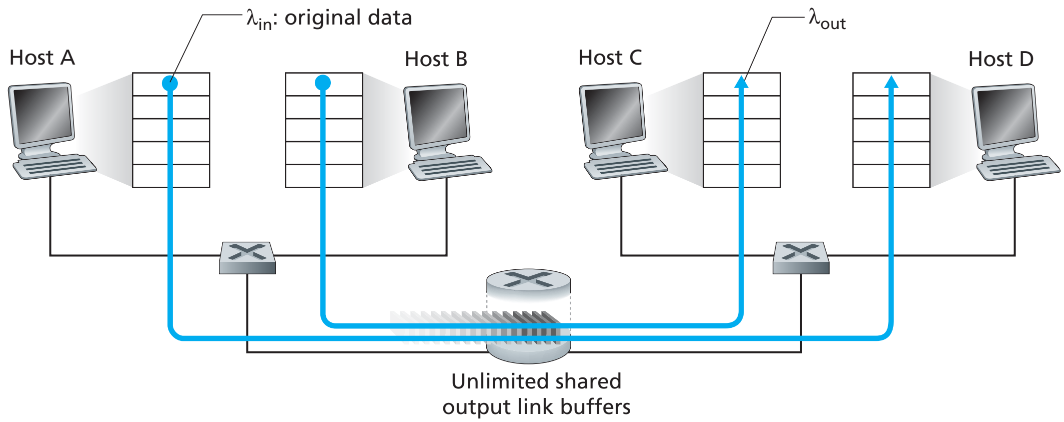

Figure 3.43 ♦ Congestion scenario 1: Two connections sharing a single hop with infinite buffers

simple one. Data is encapsulated and sent; no error recovery (for example, retransmission), flow control, or congestion control is performed. Ignoring the additional overhead due to adding transport- and lower-layer header information, the rate at which Host A offers traffic to the router in this first scenario is thus λ_{in} bytes/sec. Host B operates in a similar manner, and we assume for simplicity that it too is sending at a rate of λ_{in} bytes/sec. Packets from Hosts A and B pass through a router and over a shared outgoing link of capacity R. The router has buffers that allow it to store incoming packets when the packet-arrival rate exceeds the outgoing link's capacity. In this first scenario, we assume that the router has an infinite amount of buffer space.

Figure 3.44 plots the performance of Host A's connection under this first scenario. The left graph plots the **per-connection throughput** (number of bytes per second at the receiver) as a function of the connection-sending rate. For a sending rate between 0 and $R/2$, the throughput at the receiver equals the sender's sending rate—everything sent by the sender is received at the receiver with a finite delay. When the sending rate is above $R/2$, however, the throughput is only $R/2$. This upper limit on throughput is a consequence of the sharing of link capacity between two connections. The link simply cannot deliver packets to a receiver at a steady-state rate that exceeds $R/2$. No matter how high Hosts A and B set their sending rates, they will each never see a throughput higher than $R/2$.

Achieving a per-connection throughput of $R/2$ might actually appear to be a good thing, because the link is fully utilized in delivering packets to their destinations. The right-hand graph in Figure 3.44, however, shows the consequence of operating near link capacity. As the sending rate approaches $R/2$ (from the left), the average delay becomes larger and larger. When the sending rate exceeds $R/2$, the

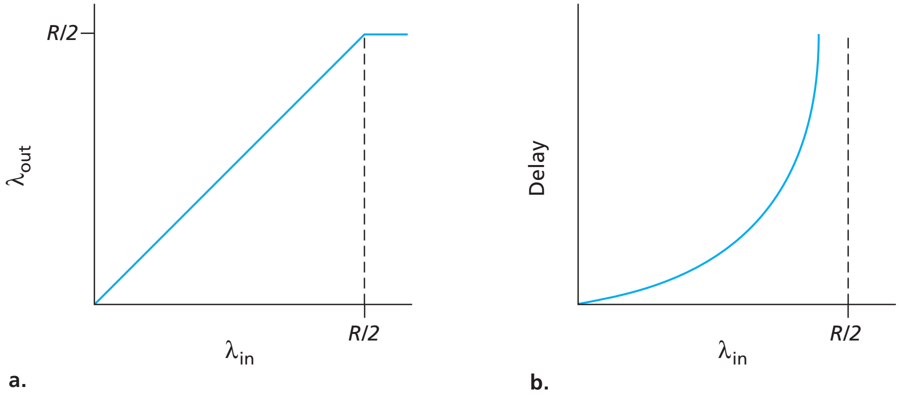

Figure 3.44 ♦ Congestion scenario 1: Throughput and delay as a function of host sending rate

average number of queued packets in the router is unbounded, and the average delay between source and destination becomes infinite (assuming that the connections operate at these sending rates for an infinite period of time and there is an infinite amount of buffering available). Thus, while operating at an aggregate throughput of near R may be ideal from a throughput standpoint, it is far from ideal from a delay standpoint. *Even in this (extremely) idealized scenario, we've already found one cost of a congested network—large queuing delays are experienced as the packet-arrival rate nears the link capacity.*

Scenario 2: Two Senders and a Router with Finite Buffers

Let us now slightly modify scenario 1 in the following two ways (see Figure 3.45). First, the amount of router buffering is assumed to be finite. A consequence of this real-world assumption is that packets will be dropped when arriving to an already-full buffer. Second, we assume that each connection is reliable. If a packet containing a transport-level segment is dropped at the router, the sender will eventually retransmit it. Because packets can be retransmitted, we must now be more careful with our use of the term *sending rate*. Specifically, let us again denote the rate at which the application sends original data into the socket by λ_{in} bytes/sec. The rate at which the transport layer sends segments (containing original data *and* retransmitted data) into the network will be denoted λ'_{in} bytes/sec. λ'_{in} is sometimes referred to as the **offered load** to the network.

The performance realized under scenario 2 will now depend strongly on how retransmission is performed. First, consider the unrealistic case that Host A is able to somehow (magically!) determine whether or not a buffer is free in the router and thus sends a packet only when a buffer is free. In this case, no loss would occur,

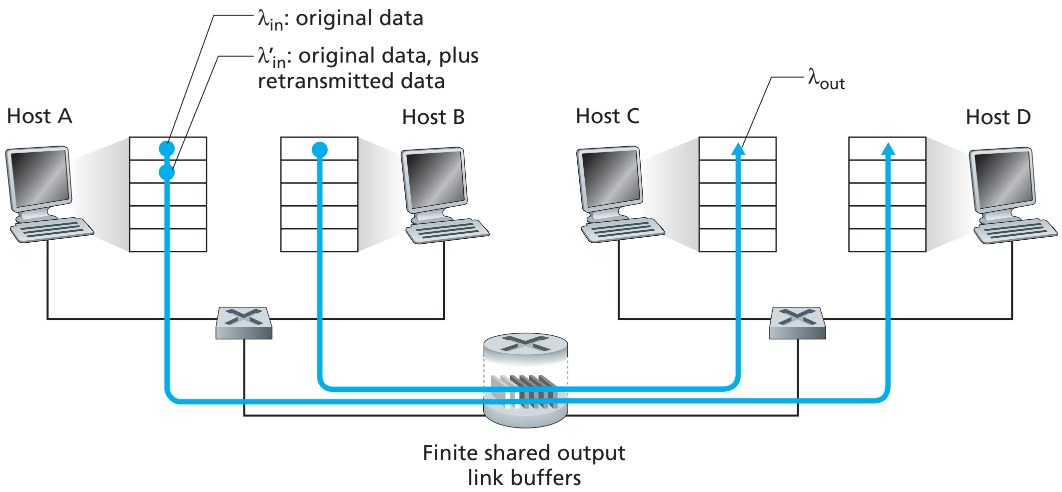

Figure 3.45 ♦ Scenario 2: Two hosts (with retransmissions) and a router with finite buffers

λ_{in} would be equal to λ'_{in}, and the throughput of the connection would be equal to λ_{in}. This case is shown in Figure 3.46(a). From a throughput standpoint, performance is ideal—everything that is sent is received. Note that the average host sending rate cannot exceed $R/2$ under this scenario, since packet loss is assumed never to occur.

Consider next the slightly more realistic case that the sender retransmits only when a packet is known for certain to be lost. (Again, this assumption is a bit of a stretch. However, it is possible that the sending host might set its timeout large enough to be virtually assured that a packet that has not been acknowledged has been lost.) In this case, the performance might look something like that shown in Figure 3.46(b). To appreciate what is happening here, consider the case that the offered load, λ'_{in} (the rate of original data transmission plus retransmissions), equals $R/2$. According to Figure 3.46(b), at this value of the offered load, the rate at which data are delivered to the receiver application is $R/3$. Thus, out of the $0.5R$ units of data transmitted, $0.333R$ bytes/sec (on average) are original data and $0.166R$ bytes/sec (on average) are retransmitted data. *We see here another cost of a congested network—the sender must perform retransmissions in order to compensate for dropped (lost) packets due to buffer overflow.*

Finally, let us consider the case that the sender may time out prematurely and retransmit a packet that has been delayed in the queue but not yet lost. In this case, both the original data packet and the retransmission may reach the receiver. Of

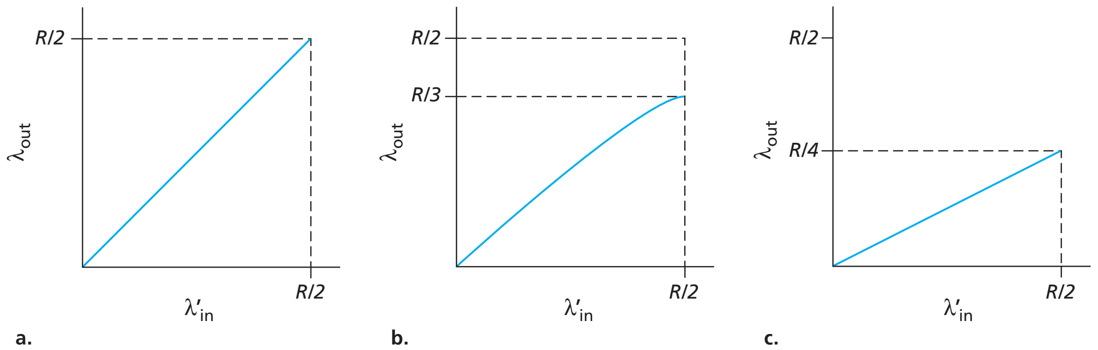

Figure 3.46 ♦ Scenario 2 performance with finite buffers

course, the receiver needs but one copy of this packet and will discard the retransmission. In this case, the work done by the router in forwarding the retransmitted copy of the original packet was wasted, as the receiver will have already received the original copy of this packet. The router would have better used the link transmission capacity to send a different packet instead. *Here then is yet another cost of a congested network—unneeded retransmissions by the sender in the face of large delays may cause a router to use its link bandwidth to forward unneeded copies of a packet.* Figure 3.46 (c) shows the throughput versus offered load when each packet is assumed to be forwarded (on average) twice by the router. Since each packet is forwarded twice, the throughput will have an asymptotic value of $R/4$ as the offered load approaches $R/2$.

Scenario 3: Four Senders, Routers with Finite Buffers, and Multihop Paths

In our final congestion scenario, four hosts transmit packets, each over overlapping two-hop paths, as shown in Figure 3.47. We again assume that each host uses a timeout/retransmission mechanism to implement a reliable data transfer service, that all hosts have the same value of λ_{in}, and that all router links have capacity R bytes/sec.

Let's consider the connection from Host A to Host C, passing through routers R1 and R2. The A–C connection shares router R1 with the D–B connection and shares router R2 with the B–D connection. For extremely small values of λ_{in}, buffer overflows are rare (as in congestion scenarios 1 and 2), and the throughput approximately equals the offered load. For slightly larger values of λ_{in}, the corresponding throughput is also larger, since more original data is being transmitted into the

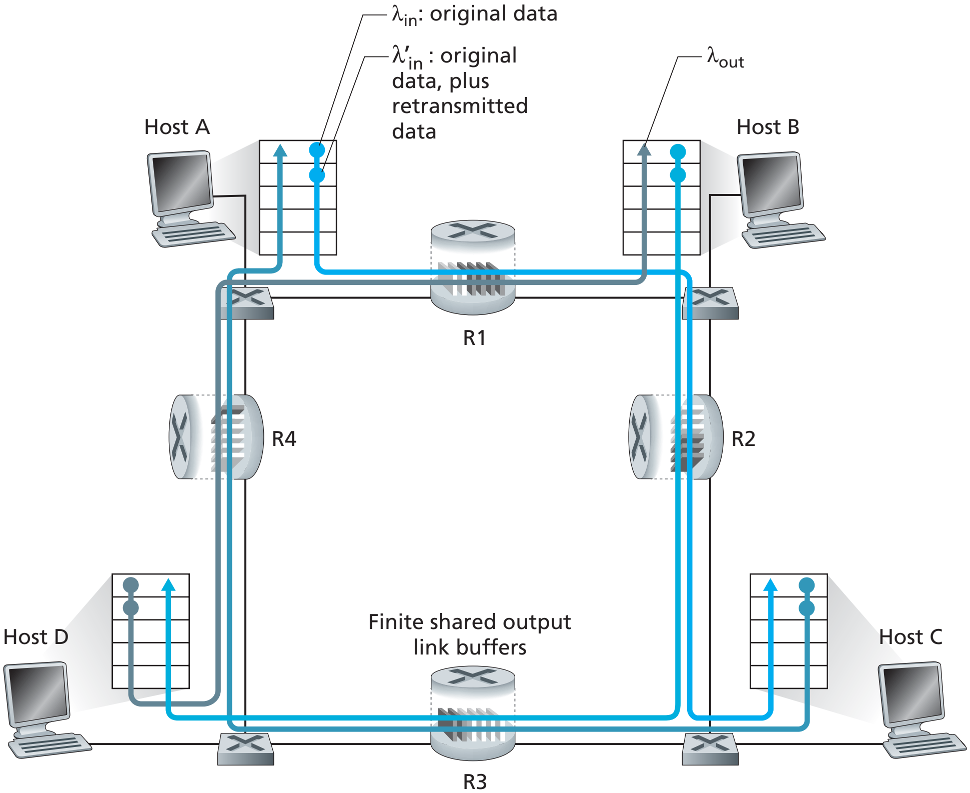

Figure 3.47 ♦ Four senders, routers with finite buffers, and multihop paths

network and delivered to the destination, and overflows are still rare. Thus, for small values of λ_{in}, an increase in λ_{in} results in an increase in λ_{out}.

Having considered the case of extremely low traffic, let's next examine the case that λ_{in} (and hence λ'_{in}) is extremely large. Consider router R2. The A–C traffic arriving to router R2 (which arrives at R2 after being forwarded from R1) can have an arrival rate at R2 that is at most R, the capacity of the link from R1 to R2, regardless of the value of λ_{in}. If λ'_{in} is extremely large for all connections (including the B–D connection), then the arrival rate of B–D traffic at R2 can be much larger than that of the A–C traffic. Because the A–C and B–D traffic must compete at router R2 for the limited amount of buffer space, the amount of A–C traffic that successfully gets through R2 (that is, is not lost due to buffer overflow) becomes smaller and smaller as the offered load from B–D gets larger and larger. In the limit, as the offered load approaches infinity, an empty buffer at R2