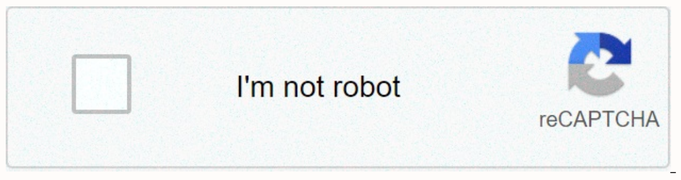

Continue

Tcp reno fast recovery

This section discusses how TCP manages clogging, both for the connection's own benefit (to improve its interriap) and for the benefit of other connections (which may cause our connection to reduce its own streaming time). Early work on overwork was a tcp-flavored climax in 1990, known as TCP Reno. Mechanisms for managing clogged TCP Reno measures remain the dominant approach on the Internet today, although alternative TCP is an active field of research and will be taken into account in 22 new TCP implementations. The central TCP mechanism here is to link to adjust the window size. Less winsize means there are fewer packages on the Internet at any time, and less traffic less congestion. Greater winsize means better flow, to the point. All TCP reduce winsize when the clogging is obvious, and increase it when it is not. The trick is in finding out when and by how much to make these winsize changes. Many improvements for TCP have come from mining more and more information from the ACK return flow. Recall chiu and Jaa's definition from 1.7 The blockage that the knee clogging occurs when the waiting line first starts to grow, and cliffs of clogging occur when the packets start to be discarded. Clogging can be managed at both points, although laid-off packages can be an important resource-efficient batch. Some of TCP's newer strategies try to take action at a clogged knee (22.6 TCP Vegas), but TCP Reno has a cliff-based strategy: packages must be lost before the sender reduces the window size. In 25 Quality of Service we will examine some router-centric alternatives to TCP to manage Internet clogging. But for the most part, these have not been universally accepted, and TCP is all that stands in the way of internet congestive collapse. The first question you could ask about TCP ingestion management is just how did he get this job? TCP sender is expected to monitor its download speed in order to work with other senders to reduce general clogging between routers. While part of each TCP's goal is good, stable performance for its own connections, this emphasis on end-user collaboration introduces the possibility of cheating: the host could be tempted to increase the flow of own connections at the expense of others. The layout of TCPs, which are tasked with clogging between core routers, is to some extent as if the handcuffs were placed in the henhouse. More specifically, such an arrangement has the potential to lead to a tragedy in common. Multiple TCP senders share a common resource — an Internet backbone — and while the backbone is most effective when each sender participates, each individual sender can improve their position by sending faster than allowed. One of the arguments used by the to support the implementation of a wide range of clogging options under the supervision of the central authority. Nevertheless, TCP has been quite successful in distributed overflating. This was partly because system vendors have an incentive to take a look at the big pictures, but in the past it has been quite difficult for individual users to replace their TCP piles with the running versions. Another contributing factor to TCP's success is that most bad TCP behavior requires collaboration at the end of the server, and most server managers have an incentive to collaborate. Servers generally want to fairly distribute bandwidth between their multiple clients, and – theoretically at least – ISP servers can punish mineath. So far, at least, the TCP approach has worked remarkably well. TCP clogging management is based on window-based; TCP adjusts the window size to adjust the clogging. The window size can be mentioned as the number of packets there on the network; more accurately, represents the number of packages and ACK either in transit or in a row. An alternative approach often applied to real-time systems is to manage clogging based on the level of accident if the shipping speed instantly exceeds the available level. In the most influential days of TCP, the size of the TCP connection window came from the AdvertisedWindow value suggested by the receiver, essentially representing how many buffer packets could be assigned. This value is often quite large, for installing large bandwidth×deley products, and so it is often reduced out of worry for clogs. When winsize is adjusted downwards, it is generally called The Block window or cwnd (the name of the variable that first appears in Berkeley Unix). Strictly speaking, winsize = min(cwnd, AdvertisedWindow). For newer TCP versions, the cwnd variable can actually be used to estimate the sender's number of packets in a year; see sidebar on 19.4 TCP Reno and Fast Recovery. If TCP sends over a wireless network, RTT will be on rtt RTTnoLoad, the travel time without delays in the waiting list. As we saw in 8.3.2 RTT Calculations, (RTT−RTTnoLoad) is the time each package spends in the queue. The route bandwidth is winsize/RTT, so the number of packets in the winsize × (RTT−RTTnoLoad) / RTT. Normally, all packets in the queue on the router on the connection header are a bottleneck. Note that the sender can calculate this number (assuming we can rate RTTnoLoad; the most common approach is to assume that the smallest measured RTT corresponds to RTTnoLoad). TCP self-training (that is, that is, that new downloads continue with the return of ACKs) ensures that, again assuming an otherwise flawless network, the waiting Self-command means that the speed of packet downloads is equal to the available bandwidth of the bottleneck. When sending burst packets, there are a few spikes (e.g. when the sender increases the size of the window), but in the self-command state, the packets accumulate only in the bottleneck. In the next chapter, in 20.2 Bottlenecks Links with the Competition, we will return to the case of a network that is not otherwise flawless. The optimal window size for a TCP connection would be the bandwidth × RTTnoLoad. With this window size, the sender accurately filled the transit capacity along the route to the destination and did not use any queue capacity. Actually, TCP Reno doesn't do that. Instead, TCP Reno does the following: guessing on the reasonable initial size of the window, Using the survey format is stacked to increase the size of the window, u slucaju that dodju dodju to loss, on the theory that the maximum in-hodjivput must not be dohvacen, brze is shredding and drugation, on the theory that, u slucaju that loss is necessary drastic action in practice, this in practice the static TCP size is ossued significantly above the theory optimum. One explanation of the TCP approach is that there are different ceilings in time on the number of packets that the network can accept. Every sender tries to stay close, but just below that level. Occasionally the sender will overshoot and the package will be dumped somewhere, but it only teaches the sender a little more about where the network ceiling is. More formally, this ceiling represents the largest cwnd, which does not lead to the loss of the package, i.g. cwnd, which at this particular moment fully fills but does not exceed the waiting run of the bottleneck. We reached the ceiling when the waiting table was full. In Chi and Jain's terminology, there's a cliff on the far side of the ceiling where the bends are lost. TCP tries to stay above the knee, which is the point when the waiting line first starts being applied persistently, making the waiting line at least partially occupied; whenever it sends too much and falls off a cliff, it retreats. The ceiling concept is often useful, but not necessarily as precise as it might sound. If we reach the ceiling by gradually expanding the size of sliding windows, then winsize as much as possible. However, if the sender suddenly omits the concept of packages, the waiting table can be filled and we will reach a temporary ceiling without fully exploiting the transit capacity. Another source of ceiling unambiguity is that the connection of the bottleneck can be shared with other connections, in which case the ceiling represents a specific proportion of our connection, which can fluctuate greatly over time. Finally, at the point where the ceiling is reached, the queue is full, so a significant number of packages are waiting in the queue; sender cannot drag back It is time to acknowledge the existence of different versions of TCP, which include different clogging algorithms. For starters, TCP Tahoe (1988) and TCP Reno (1990) will start; The names Tahoe and Reno were originally code names of berkeley unix distributions, which included these relevant TCP implementations. The ideas for TCP Tahoe came from the 1988 Jacobson and Karelsa newspapers; TCP Reno was then cleared a few years later. TCP Reno is still in widespread use more than twenty years later and is still an undeniable reference implementation of TCP, although some modest improvements have been made (NewReno, SACK). A common theme for the development of improved TCP implementations is that one end of the connection (usually the sender) extracts larger and larger amounts of information from the packet flow. TCP Tahoe, for example, introduced the idea that duplicate ACDs possibly represent a lost package; TCP Reno introduced the idea that reverse duplicate ADs are linked to packets that were successfully sent but tracked by loss. TCP Vegas (22.6 TCP Vegas) has introduced fine grainy RTT measurement in order to detect when RTT > RTTnoLoad. It is often useful to think that the TCP sender has breaks between consecutive window windows; this means that the sender sends CWND packets, is briefly insignless, and then sends other cwnd packets. Successive packet windows are often called flights. However, the existence of any separation between the years is not guaranteed. We will begin with a situation in which TCP has established a reasonable guess for the cwnd, comfortably below the size of the advertised window, and which largely indicates that it works. TCP then deals with some fine tuning. This TCP stock situation – in terms of regular fluctuations – is usually called the clogging avoidance phase, although all stages of the process are ultimately aimed at avoiding clogging. The central strategy is that when the package is lost, the cwnd should be reduced quickly, but otherwise it should increase slowly. This leads to slow fluctuations in cwnd, which over time allows the average cwnd to adapt to long-term changes in network capacity. When TCP completes each packet window, it determines whether there has been a loss. The CWND adjustment rule introduced by TCP Tahoe and [JK88] is as follows: if there were no losses in the previous window, cwnd = cwnd +1 if the turns were lost, cwnd = cwnd/2 Informally dose cwnd in full-pack units; strictly speaking, cwnd is measured in bytih and is increased by the maximum size of the TCP segment. This strategy is known as additive enhancement, multiplier reduction or AIMD; cwnd = cwnd+1 is an idetive increase and cwnd = cwnd/2 is a multiplier reduction. Normally, the setting cwnd=cwnd/2 is a medium-term goal; actually, TCP Tahoe cwnd=1 immediately after the actual time-lapsed. Without loss, TCP will send consecutive windows, e.g. 20, 21, 22, 23, 24, ... That means conservative network probing and especially waiting in a bottleneck router. TCP tries to increase cwnd values because the absence of loss means that the current cwnd is below the network ceiling; this means that the waiting order on the router bottleneck is not too much. If loss occurs (including multiple losses in one window), TCP's response is to reduce the window size in half. (As we'll see, TCP Tahoe actually treats it in a somewhat circular way.) Informally, the idea is that the sender must respond aggressively to the clogging. More specifically, lost packets mean that the queue of the narrow router is filled, and the sender must dial back to the level that will allow the waiting order to be cleared. Assuming that transit capacity is approximately equal to the wait line capacity (they say each equals N), then we overw but drop the packets when cwnd = 2N, leaving us cwnd =cwnd/2 with cwnd =N, which only fills the transit capacity and leaves the queue blank. (When the consignor lists cwnd=N, the actual number of packets in transit takes at least one RTT to fall from 2N to N.) Of course, assuming that any relationship between transit capacity and queue capacity is very speculative. At a 5,000 km optical connection with a 10 Gbps bandwidth, a round transit capacity would be about 60 MB or 60,000 1kB packets. Most routers probably don't have such large waiting routes. However, the total capacity is in the waiting order beyond transit capacity. On the other hand, according to the TCP high bandwidth model, a small part of the bandwidth is a small part of the ×delay product. We return to this in 19.7 TCP and Bottleneck Link Utilization and 21.5.1 Bufferbloat. Note that if TCP is loss of the packet and there is an actual time clock (as opposed to the loss of the package detected by Fast Retransmit, 19.3 TCP Tahoe and Fast Retransmit), the sliding window tube has dried out. There's no package on the plane. No self-command can edit new downloads. Sliding windows must therefore restart from scratch. The stopper avoidance algorithm leads to the classic TCP sawtooth graph, where peaks are at points where the slowly rising cwnd crosses over the network ceiling. We emphasize that TCP sawn teeth is specific to TCP Reno and related TCP implementations that share the Reno mechanism to increase additives/multiplier reductions. In periods without loss, TCP cwnd lineano increases; when losses occur, TCP specifies cwnd = cwnd/2. This diagram is idealization, as when a loss occurs, it takes the sender some time to discover it, perhaps as much as the timeOut interval. Oscillation shown here in the red ceiling somewhat arbitrary. If there is only one or two other opposite consignors, the deviation of the ceiling can be quite dramatic, but for many consignors, the changes can be smoothed out. For some TCP sawing graphs created by the actual simulation, see 31.2.1 Cwnd Graph at a time and 31.4.1 Some TCP Reno cwnd graphs. The transit capacity of the route is more or less relentless, as is the physical capacity of the waiting dose on the router of the bottleneck. However, these capabilities are also shared with other connections that can come and go with time. That's why the ceiling differs in real terms. If two other connections share a path with a total capacity of 60 packages, it can be the fairest allotment for each connection to get about 20 packages as its share. If one of these connections is disconnected, the remaining two can be increased to 30 packets. And if, instead of a fourth link, joins the mix, then after the balance is reached, each link can be hoped for a fair share of 15 packages. Will such a fair allocation actually happen? Or can one connection end up getting 90% bandwidth, while two others get 5%? Chiu and Jain [CJ89] showed that the additive/multiplier reduction algorithm actually converge into approximately the same bandwidth division, If the two connections have a common bottleneck connection, provided that both ties have the same RTT u dan to RTT, it was that the two connections experienced the loss of the package, or the threads to see it, some cwnd1 i cwnd2 zaustave-window sizes, i consider the amount of cwnd1 − cwnd2. For all RTT in which there is no loss, cwnd1 and cwnd2 both overgrow by 1, and so cwnd1 − cwnd2 stays the same. If there is a loss, then both are cut in half and so the cwnd1 − cwnd2 is cut in half. Thus, over time, the original value of the cwnd1 - cwnd2 is cut several times in half (during each RTT in which losses occur) until it is ed-by-case, cwnd1 ≃ cwnd2. Graphical and tabular versions of this same argument are in the following chapter in 20.3 TCP Reno Fairness with synchronized losses. The second hypothesis mentioned above can be called the synchronized loss hypothesis. While it is very reasonable to assume that these two links will experience the same number of losses as the long-term average, it is a much stronger statement that assumes that all loss events are shared between the two connections. This behaviour may not happen in real life and has been the subject of some debate; see [GV02]. At this point we return to 31.3 Two TCP senders that expire. Fortunately, RTT's equality is still valid if each connection is equally likely to experience the loss of the package: both connections will have the same level of loss, and so, as we will see in the 21.2 TCP Reno loss rate compared to cwnd, will have the same loss rate. honesty can take much longer. In 20.3 TCP Reno Fairness with synchronized losses we also look at some alternative hypotheses for the case of unequal-RTT. How do we speculate on network capacity? What value should we start with? And even if we have a good target for CWND, how do we avoid flooding the network that sends the initial burst of packets? The answer is known as a slow start. If you try to guess a number in a fixed range, you'll probably use binary search. Not knowing the volume behind the network ceiling, a good strategy is to guess cwnd=1 (or cwnd=2) at the beginning and continue to duplicate until you have gone too far. Then, after a previous guess, which is known to have worked. At the moment, you are assured that you are within 50% of the right capacity. The actual slow-start mechanism is to increase the cwnd for each ACK received by 1. This seems like a lineano, but it's misleading: after we send window packets (cwnd a lot), we received cwnd ACKs and so they have inkremented cwnd-many times, and so they are set cwnd to (cwnd+cwnd) = 2×cwnd. In other words, cwnd=cwnd×2 after each window is the same as cwnd+=1 after each package. Assuming that packages travel together in the window window, all this means cwnd doubles each RTT during a slow start; This is probably the only place in computer literature where exponential growth is described as slow. However, it is really slower than the alternative to sending an entire window at once. Here's a slow start chart. This diagram makes it possible to make the implicit assumption that the load-free RTT is large enough to hold well over 8 packages of the largest window size shown. For another example, with a much smaller RTT, see 19.2.3 Slow-Start Multiple Drop Example. Eventually, the waiting order of the bottleneck fills up and the package is dumped. Let's say it's after N RTTs, so cwnd=2N. Then, during the previous RTT, cwnd=2N-1 worked successfully, returning to that previous value by setting cwnd =cwnd/2. During slow start, increasing cwnd by one per inbox ACK is equivalent to duplication of cwnd after each window. A similar equivalence can be found for the stopper avoidance phase, above. While avoiding clogging, the cwnd after each window zoom from 1. To formulate this as an increase on the ACK, we extend this prirachek 1 throughout the window, which of course has the size of a cwnd. This is next with each ACK received: It's a slight approximation as the cwnd continues to change, but it works well in practice. Since TCP actually measures cwnd in bytes, floating point arithmetic is not usually required; see 14.0. The exact equivalent of the strategy for zooming on the window is cwnd = cwnd + 1/cwnd0, where the cwnd0 value is cwnd at the beginning of this specified window. Another, simpler approach is to cwnd += 1/cwnd, and that part will be recorded, but use the soil (cwnd) (integer part cwnd) when actually sending packets. Most actual implementations are accompanied by cwnd in bytes, in which case the use of an even number of arithmetic is sufficient until the cwnd becomes quite large. If a delayed AKK (18,8 TCP Delayed ACKS) is implemented, then in bulk one that arrives ACK actually recognizes two packages. RFC 3465 allows the TCP receiver to run cwnd for 2/cwnd in this position, which is the answer corresponding to increasing the cwnd by 1 after receiving enough ACK to confirm the entire window. Sometimes TCP uses slow startup even when it knows the work network capacity. After losing the package and the timing of the TCP, he knows that the new cwnd cwnd cwndold/2 must work. If it were cwnd 100, TCP halves it to 50. The problem is that after the time shut down, no ACKs are returned to self-download, and we don't want to boil 50 packets into the network all at once. So when restarting

the TCP stream again uses what may be called a slow start threshold: it uses slow startup, but stops when cwnd reaches its destination. Specifically, on the loss of the package we determine the variable ssthresh on cwnd/2; This is our new target for CWND. Set the cwnd to 1, and switch to slow start mode (cwnd += 1 for each ACK). However, when cwnd reaches ssthresh, switch to clogging avoidance mode (cwnd += 1/cwnd for each ACK). Note that switching from a threshold to a slow start to avoid clogging is completely natural and easy to implement. TCP will use the slow start threshold when it is restarted from the pipe drain; this is, every time a slow start is required after its first use. (If the connection was simply in a regular way, slow-running without a threshold is usually used when traffic restarts.) Threshold slow-start can be seen as an attempt to combine the rapid expansion of windows with self-commanding. By comparison, we can refer to the initial, boundless slow start as a non-limiting slow start. Note that a boundless slow start serves a significantly different purpose – initial probing to set a network ceiling within 50% – from the slow start limit. Here is the TCP saw diagram above, modified to display timeouts and slow start. The first two losses of the package are shown as extensive time evasiveness; the rest are shown as if fast retransmit had been used, below. RFC 2581 makes a slow start with cwnd=2. A slow start has the ability to cause more dropped packets on the bottleneck connection; loss of packets for quite some time because TCP sender is slowly discovering them. The network topology is as follows, where the A-R connection is infinitely fast, and the R-B connection has a bandwidth in the R⟶B 1 packet/ms direction. ACKS travel immediately from B back to A. In this and subsequent cases, we will continue to use the data[N]/ACK[N] terminology 8.2 of the Windows sliding system, ing with N=1; TCP numbering is not done in this way, but the distinction is non-clinical. When A uses slow start here, the sequential windows will almost immediately start to overlap. A will send one packet to T=0; will be delivered to T=1. ACK will travel immediately to A, at which point A will send two packages. From now on, ACK will come regularly at a rate of 1 per second. Here is a short chart: Time A receives R sends R's queue 0 Data[1] Data[1] 1 ACK[1] Data[2],Data[3] Data[2] Data[3] 2 ACK[2] 4.2 2 000 5 4.5 3 ACK[3] 6.7 4 5.7 4 ACK[4] 8.9 5 6.9 5 ACK[5] 10.11 6 7.11 .. N ACK[N] 2N,2N+1 N+1 N+2 .. 2N+1 For T=N, R's waiting order contains N packets. At T=100, the R's waiting table is full. The data [200] sent to T=100 will be delivered and validated to T=200, which will give it 100 RTT. At T=101 R, it receives Data[202] and Data[203] and drops the latter. Unfortunately, the time interval A must of course be greater than RTT, so A will not detect loss until, at least, T=200 detects loss. At the moment, A sent packets through data[401], 100 Data packets,[203], Data[205], ..., and Data[401] were all lost. In other words, at the point where A first receives the news of one lost package, in fact at least 100 packages have already been lost. Fortunately, boundless slow momentum generally occurs only once per connection. TCP Tahoe has one important function. Remember that TCP ACK is cumulative; if packages 1 and 2 have been received and data arrives now[4], but still data[3], all they can (and must!) do is send back another ACK[2]. Thus, from the sender's point of view, if we send packages 1,2,3,4,5,6 and return ACK[1], ACK[2], ACK[2], ACK[2], ACK[2], ACK[2], we can write down two things: Data[3] is lost, So we stuck to ACK[2] Data 4.5 and 6 probably did make through, and triggered the three duplicate ACK[2]s (the three ACK[2]s following the first ACK[2]s. The quick transfer strategy is to resend data [N] when we receive three dupACK data[N-1]; i.e. four ACK[N-1]. Since this represents the loss of the package, we also determine ssthresh = cwnd/2, place cwnd=1 and start the threshold-slow-start phase. The effect of this is usually to reduce the delay associated with the lost package, from the full time lag, usually 2×RTT, to little more than one RTT. The lost package is discovered before the TCP pipeline is outflow. However, at the end of the next RTT, when the TCP pipeline is returned, the TCP pipeline will be ooze, so it needs to start slowly. TCP Tahoe included all the functions discussed so far: cwnd+=1 and cwnd=cwnd/2, slow start, and Quick A quick retransmit waits for the third dupACK, which allows the possibility of moderate oversumerment of packages. Let's say packages 1 to 6 are sent, but they come to order 1,3,4,2,6,5, perhaps because of a router on the go with an architecture that is strongly parallel. Then the acki that would be sent back, Next: Received response data[1] ACK[1] Data[3] ACK[1] Data[4] ACK[1] Data[2] ACK[4] Data[4] Data[4] Data[1] Data[4] Data[4] Data[4] Data[4] Data[4] Data[4] 4] 6] ACK[4] Data[5] ACK[6] Waiting for the third dupACK is in most cases a successful compromise between responsiveness to lost packets and reasonable records that the data packet is actually lost. However, a router that makes a more substantial delivery redirection would destroy the dent on the connections using a quick retransmict. In particular, note the router R in the diagram below; when sending packets to B, it could in principle be to be replaced on the basis of a package by package between the route via R1 and the route via R2. That would be a mistake; If R1 and R2 routes had different delays in reproduction, this strategy would introduce a major retraining of packages. R must send all turns belonging to any TCP connection by one path. In the real world, routers generally go to considerable lengths that could be quickly retransmit; in particular, the use of multiple paths for a single TCP connection is almost universally frozen. Some actual information about the examination of packages can be found in [VP97]; the author suggests that switching to retransmission to another dupACK would be risky. A quick retransmit requires the sender to set CWND=1 because the pipe has been oded and no ACDs for download speed are received. Speedy recovery is a technique that often allows the sender to avoid drying out the pipe, and to move from cwnd to cwnd/2 in the space of one RTT. TCP Reno is TCP Tahoe with the addition of Fast Recovery. The idea is to accelerate retransmission with the arrival of dupacks. We assume that each received dupack indicates that some data packet after the lost packet was successfully delivered; Turns out it doesn't matter which one. When the lost package was discovered via The Rapid Retransmit, we set up cwnd=cwnd/2; The next step is to figure out how many dupacks we have to wait before we can continue the transfer of new data. At the very least, at least we assume that only one data package is lost, although in the next section we will see that more losses can be dealt with by a slight change in the rapid recovery strategy. During the recovery process, we cannot directly use cwnd because the scroll window cannot scroll until the lost package is downloaded. Instead, we will use the concept of Estimated FlightSize or EFS, which is the best guess of the sender in terms of the number of exceptional packages. Under normal circumstances, the EFS is the same as cwnd. A key observation of rapid recovery is that the EFS should be reduced by 1 arrives dupACK. First we outline the general case, and then we look at a specific case. Let cwnd = N, and assume that package 1 is lost (packet numbers can be taken as relative). Until package 1 is re-sent, the sender can only send up via the N package (Data[N] can only be sent after ACK[0] has arrived per sender). The receiver will send N−1 dupACK[0]s representing packages 2 to N. At the point of the third dupACK, when the data loss was detected[1], the sender calculates as follows: EFS was cwnd = N. Three dupACKI have arrived, representing three subsequent packages that are no longer in a year, so the EFS is now N−3. At this point, the sender determines that the package has been lost, making the EFS = N−4 short, but this package is immediately transferred, which returns the EFS to N−3. The sender expects to receive an N−4 dupACK at this time, and another new ACK to retransmission the lost package. This last ACK will be for the entire original window. The new target for CWND is N/2 (for simplicity we will assume that N is even). So, we're waiting for N/2 to arrive − another 3 dupACK, at which point the EFS is N−3−(N/2−3) = N/2. After this point, the sender will continue to send new packages; will send one new package for each of the N/2−1, which will then arrive dupACK (remember that N−1 dupACK in all). These new transfers will be data[N+1] via data [N+(N/2−1)]. After the last of the dupACK will come ACK, which corresponds to the retransmission of the lost package; will be an ACK[N] that recognizes all the original window. At this point, the N/2 − 1 unknown packets of Data[N+1] via data [N+(N/2)−1. The sender sends data[N+N/2] to continue scrolling windows with cwnd = N/2: The sender has received ACK[N] and has exactly one full window open for the new N/2 value from cwnd. That means we're right where we should be. Here's a diagram showing the quick recovery for cwnd=10. Data[10] is lost. Data[9] are eligible for the initial ACK[9] and nine Data packages[11] through data[19] are each extracted from dupACK[9]. Denote the dupACK[9] from data[N] from dupACK[9]/N; these are shown along the upper right. If SACK TCP (below) is not used, the sender will not be able to identify the N or distinguish these dupACK apart. When the sender arrives at the consignor dupACK[9]/13 (third dupACK), the sender uses quick recovery to understand that the data[10] is lost and reassigns it. At this point EFS = 7: the sender sent the original serojo 10 data packets, plus Data[19], and received one ACK and three dupACK, a total of 10+1-1-3 = 7. The sender also considered that data[10] was lost (EFS −= 1) and then re-downloaded it (EFS += 1). There are six more dupack[9] on the way. THE EFS is reduced for each subsequent arrival of dupACK; when we get two more dupACK[9]'s, efs is 5. V dupACK[9] (dupACK[9]/16) reduces the EFS to 4 and thus allows us to transmit Data[20] (which immediately encounters the EFS back to 5). We have received a send dupACK[9]/16 Data[20] dupACK[9]/17 Data[21] dupACK[9]/18 Data[22] dupACK[9]/19 Data[23] We reiterate that the TCP sender does not see the numbers 16 to 19 in the above receipt column; specifies when to start the number of arrivals of dupACK[9]. The next thing that will come to the sender's page is the ACK[19] provided by the data[10]; at the point where the data [10] arrive at the receiver, the data[11] has already arrived through the data[19] and thus the cumulative-ACK answer is the ACK[19]. The sender is responding to ACK[19] with data[24] and the transition to cwnd=5 is now complete. During lossless scrolling windows, the sender will send CWND packets to RTT. If a large time printout occurs, it is usually detected only after at least one full RTT timeless connection; additionally untapped RTT during the slow start-up phase. It is worth studying the rapid recovery sequence shown in the illustration in terms of untapped bandwidth. The diagram shows the three times of the encircle the journey, as seen from the sender's page. Ten Data packets[9]-Data[18] are sent during the first RTT. The second RTT starts with the transmission of data[19] and continues with the transmission of data[22], together with the data used[10]. The third RTT starts with the transmission of data[23] and includes through data[27]. As regards the effectiveness of recovery, RTT sends 9, 5 and 5 packages (data were counted twice[10]); this is extremely close to the ideal of reducing cwnd to 5 immediately. The reason that we cannot use cwnd directly in the rapid recovery formulation is that, until the lost data[10] is acknowledged, the window is frozen at data[10]-Data[19]. The original RFC 2001 description of the rapid recovery described a retransmission in terms of cwnd inflation and deflation. Inflation would start at the moment when the sender was sending new packets again, in which the cwnd would start to increase for each dupack; in the chart above, CWND would have finished at 15. When ACK[20] finally arrived from the lost package, the recovery phase would end and the CWND would immediately be flushed to 5. For a diagram showing CWND inflation and deflation, see 32.2.1 Script management. TCP NewReno described in [JH96] and RFC 2582 (currently RFC 6582) is a modest recovery process that greatly improves the handling of a case where two or more packets are lost in the window. It is considered part of the modern TCP Reno. If two data packets are lost and the first is re-sent, the receiver will confirm the data up to something before the second packet and then continue sending dupACKs of this until the second lost packet is sent back. These data acki until before the start of the second package are called partial ACK because the retransmiment of the first lost packet did not result in ACK all open data. The NewReno mechanism uses these partial ACK as evidence to retransmit later lost packages, as well as to continue the process of rapid recovery. In the diagram below, packages 1 and 4 are lost in 12-size window 0..11. Initially, the sender will get dupACK[0]'s; The first 11 ACK (right-to-left lines) are ACK[0] and 10 dupACK[0]'s. When package 1 is successfully passed on receipt of the third dupACK[0], the response of the receiver ACK[3] (heavy line on the line) will be. This is the first partial ACK (full ACK would be ACK[12]). Upon receipt of any partial ACK during the rapid recovery process, TCP NewReno assumes that the next data packet has been lost immediately and is immediately re-sent; the sender is not waiting for three dupACK, because if the next data packet had not been lost, instances of partial ACK would never have been created, even if the packet had been reused. The response of TCP NewReno sender is that each partial ACK is treated as dupACK[0], except that the sender also re-sends a data packet that – based on receiving a partial ACK – can conclude that it is lost. NewReno continues to make a quick recovery from the site, whether they are original dupack or later partial ACK or dupACK. When the first ACK of the recipient[3] arrives at the sender, NewReno reports that Data[4] has been lost and re-sent; This is the second hard data line. There is no need for dupACK[3]; as mentioned above, the consignor referred to in the single ACK[3] may prove to be lost. The sender also responds as if another dupACK has arrived[0] and sends the data[17]. The arrival of the ACK[3] indicates a reduction in the EFS of 2: one to close that Data[4] was lost, and one as if another dupACK had arrived[0]; two transmissions in response (data[4] and data[17]) return the EFS to where they were. At the moment data[16] is sent the actual (not estimated) flight is 5, not 6, because one dupACK is less[0] due to data loss[4]. However, when NewReno re-sends the data[4] and then sends the data[17], the actual flight is back by 6 March 2017. Four more dupacks arrive. NewReno sends new information about the receipt of each of this information; this is data[18] via data[21]. The receiver's response to retransmitta data[4] is to send ACK[16]; this is the cumulation of all data received up to this point. The moment this ACK comes back to the sender, he just sent Data[21] in response to the fourth dupACK[3]; its response to the ACK[16] is to send the following Data Package[22]. The sender is now back on normal sliding windows, with cwnd 6. Similar data[17] immediately after the data have been transmitted[4] qualify for ACK[17] Data [N] for fully matching ACK[N] since the beginning of losses) and the appropriate response to ACK[17] continued with the data[23]. As in the previous example of rapid recovery, we take into account the number of packages sent to RTT; The diagram shows four RTTs as shown by the sender. RTT First packet Packets sent count first Data[0] Data[0]-Data[11] 12 second Data[12] Data[12]-Data[15], Data[1] 5 third Data[16] Data[16]-Data[20], Data[4] 6 fourth Data[21] Data[21]-Data[26] 6, Re), after the loss detection, reach the new cwnd of 6 with only one packet (u second RTT). However, NewReno can only send one retransmite package to RTT. Note that TCP Newreno, like TCP Tahoe and Reno, is an innovation on the sender's side; the receiver does not need to do anything special. The following TCP flavor, SACK TCP, requires a change on the receiver side. Traditional TCP ACK is a cumulative recognition of all data received up to this point. If data[1002] is received but not Data[1001], then all the recipient can send is a duplicate ACK[1000]. This shows that something according to the data[1001] succeeded, but not more. To ensure greater specificity, TCP now provides a selective ACK (SACK) option that runs on the receiver. If this is available, the sender does not need to guess from dupACKs what he got through. The receiver can send an ACK that says: All packages up to 1000 have been received (cumulative ACK) All packages up to 1050 have been received except 1001, 1022 and 1035. The second line is part of SACK. This is supported by almost all TCP implementations. In particular, sacki include the following information; additional data that exceed the cumulative ACK are included in the TCP option field. Latest cumulative ACK Three last blocks of consecutive packages that we received if we lost 1001, 1022, 1035 and now 1051, But best of all, 1060, SACK could be said: From 1060-1052 is 1060-1052, 1050-1036 was 1034-1023 He is here, and 1001s and 1022 were not received. But nothing about packages in between. However, if the sender has been following the previous SACK received closely, he already knows that all packages from 1002 to 1021 have been received. The term SACK TCP is usually used to support selective ACKS, and the submit page is a simple TCP Reno change to take advantage of them. In practice, selective AKK at best provide modest improvement in performance in a number of situations; TCP NewReno is pretty good in an environment of moderate loss. Document [FF96] compares tahoe, Reno, NewReno and SACK TCP in cases involving one to four packet losses in one RTT. While Classic Reno performed poorly with two losses in one RTT and extremely poor with three losses, the scenarios of three losses to NewRena and SACK TCP played remarkably similarly. It was only when the connections suffered four losses in one RTT, that SACK TCP's performance began to pull slightly ahead of NewReno's performance. Consider sendering TCP Reno without traffic. As cwnd saws up and down, what happens to the throughputo? Do these cwnd halves cause at least a scaking in the stream? The answer depends to some extent on the size of the queue before the bottleneck connection, depending on the transit capacity of the route. As discussed in 8.3.2 RTT Calculations, when cwnd is less than transit capacity, the connection is less than 100% utilized and the waiting line is empty. When cwnd is more than transit capacity, the connection is saturated (100% utilized) and the waiting line has in it about (cwnd − transit_capacity) packages. The diagram below shows two TCP Reno teeth; in the first, the capacity of the waiting line exceeds the transit capacity, and in the second, the capacity of the waiting line is a much smaller part of the total. In the first diagram, the connection of the bottleneck is always 100% exploited, even on the left edge of the teeth. In the second, the interval between loss events (left and right edge of the tooth) is divided into the unsaturated connection phase and the filling phase of the queue. In an unsaturated period, the utilization of the bottleneck is less than 100% and the waiting bar is empty; At a later stage, the connection is saturated and the queue starts to charge. Rethink the idealised network below, with r-B 1 packet/≥ ms bandwidth. Assume that the total delay of RTTnoLoad is 100 ms, mainly due to a delay in reproduction; so the bandwidth×delay product is 100 packages. The issue to address is which TCP Reno, when the slow start ends, sometimes leaves the R&B connection in peace. The R-B connection will be saturated at all times provided that A always keeps 100 packets in transit, i.e. we always have a cwnd ≥ 100 (8.3.2 RTT Calculations). If cwndmin = 100, then cwndmax = 2×cwndmin = 200. To be the maximum, the waiting list capacity must be at least 99, so that the route can install 199 packages without loss: 100 packages in transit plus 99 packets in the waiting order. In general, TCP Reno never leaves a bottleneck until the queue capacity before this connection is at least as large as the running capacity for round paths. Now let's say the size of the waiting group is 49 or about 50% transit capacity. The loss of the package occurs when cwnd reaches 150, and so cwndmin = 75. Qualitatively, this example is presented with the second diagram above, although the percentage in the waiting network_ceiling 1:8 as 1:3. Now there are periods when the R-B connection is flawless. At RTT intervals, when cwnd=75, the transition will be 75% and the R-B connection will be 25% of the time in the Ned. However, CWND 75 will only be the first RTT after a loss. After 25 RTT, the CWND will be back to 100 and the connection will be saturated. So we have 25 RTTs with an average cwnd of 87.5 (=&75+100)/2), which means that the connection is 87.5% saturated, followed by 50 RTT, where the connection is 100% saturated. The long-term average here is a 95.8% utilization bottleneck link. This is not bad, given that the use of 10% bandwidth connection on warheads is almost universally considered reasonable. In addition, at the point where cwnd falls after losing to cwndmin=75, the waiting order must be full. One or two RTTs may need to drain the waiting list; in the meantime, the link utilization will be even higher. If most or all of a bottleneck connection is saturated, as in the first diagram, it can help you consider the average size of the queue. Let the waiting capacity be Cqueue and transit capacity be Ctransit, with Cqueue > Ctransit. Then the cwnd differs from maximum Cqueue +Ctransit to at least what expires (Cqueue-Ctransit)/2 + Ctransit. You would expect an average waiting group size about halfway between these, less Ctransit expression: 3/4× Waiting order - 1/4×Ctransit. If Cqueue=Ctransit, the expected average waiting group size should be approximately Cqueue/2. See exercises 12.0 and 12.5. From a connection usage point of view, the previous section suggests that the router's waiting lines are larger, not smaller. At least as large as transit capacity seems to be a great choice. To configure the router in this way, we first guess at the average RTT, and then multiply it with the output bandwidth to get the desired queue capacity. For an average RTT of 50 ms, a bandwidth of 1 Gbps leads to a waiting capacity of about 6 MB or 4000 packets per 1500 bytes each. If the numbers rise to 100 ms and 10 Gbps, the queue capacity must be 125 MB. Unfortunately, while large waiting times are useful when traffic is made up exclusively of TCP mass transmissions, they introduce relatively large waiting times that can wreak real-time traffic. Router bottleneck with the size of the waiting group corresponding to the bandwidth of the stream×delay product will double the RTT for this stream, at points when the waiting group is full. Even worse, if the target is 100% TCP connection utilization always, then the router waiting bar should be the size for the highest bandwidth flow with the maximum RTT; A short TCP connection will encounter a waiting limit much larger than necessary. This problem of large waiting capacity, which leads to excessive delay, is known as buffer; We'il be back at 9:50 Bufferbloat. Due to delays caused by large waiting times, TCP connections sometimes have to go through narrow routers with small wait times. In this case, the TCP Reno tooth connection is divided into a large phase that is not saturated on the connection small filling phase of the queue. V V for large buffers, if the target is almost 100% queue utilization, it is largely specific to TCP Reno sawtooth. Some other TCP implementations (especially TCP Vegas, 22.6 TCP Vegas) don't over-fill the queues. However, TCP Vegas does not compete well with TCP Reno, at least with the traditional FIFO queue (20.1 First look at the queue) (but see 23.6.1 Fair Queuing and Bufferbloat). The worst case scenario for using a TCP connection is if the queue size is not close to zero. Using bandwidth×delay product 100 packages, a zero waiting group will mean that the cwndmax will be 100 (or 101), so cwndmin will be 50. The utilization link therefore moves between the life of the tooth, from the low 50/100 = 50% to the high 100%; average utilisation is 75%. Although this is not ideal and while some non-reno TCP versions have tried to improve this image, 75% connection utilization is not all that bad, and can be compared to 10% bandwidth consumed as packet headers (although this figure provides 512 byte data per packet, which is low). (Literally zero waiting time won't work at all; one of the reasons – though not the only one – is that TCP Reno sends a two-pack burst when the cwnd increases.) The transport mix has a major impact on the appropriate queue size. For example, an analysis of the previous section has taken over a single long-term TCP connection. Connection usage status is improved by increasing the number of TCP connections, at least if the losses are uninhronized, as halving the cwnd of one connection has a relatively minor impact on the shared use of the queue. [AKM04] shows that for a router with N TCP connections with unsynchronized losses, the queue size (RTTaverage × bandwidth)/√N is sufficient to keep the connection almost always saturated. Higher N values are usually associated with core routers. Paper [EGMR05] suggests even smaller reflective capabilities in the order of the logarithm of the maximum window size. However, the argument makes two important assumptions: firstly, that we are prepared to tolemit the use of a link of slightly less than 100 % (albeit larger than 75%), and secondly, perhaps more importantly, that TCP is modified to extend all packet cracks – even size 2 cracks – at small intervals. There are other problems created by too many small queues, even if we are willing to accept 75% utilization of links. Internet traffic, not unlike city-bus traffic, is being theeded up; waiting as a way that these parcels do not lead to unnecessary losses. For one example of unexpected traffic, see 31.4.1.3 Transition queue peaks. Increased traffic randomisation helps reduce the need for very large waiting times, but can increase the purchasing effect. Internet core routers see more random traffic than or edge routers; in the latter are often the most difficult to configure. We will return to the question of using the connection in 31.2.6 ONE-way overput experiments and (for two senders) 31.3.10.2 Higher bandwidth and connection usage, using the NS simulator to collect experimental data. See also practice 12.0. Finally, the waiting breastfeeding capacity does not have to be static. We will return to 21.5 Active Queue Management at this point. Moreover, many of the queue problems ultimately stem from the fact that all traffic is dumped into a single FIFA queue; we will look at alternative waiting strategies in 23 queues and scheduling. For a specific case related to buffer, see 23.6.1 Fair waiting and buffer bloata. If there is no competition again on the bottleneck link, the policy of increasing TCP Reno supplements has a simple consequence: at the end of each tooth, only one packet will be lost. To see this, make it a sender, R router bottleneck, and B receiver: Keep the T bandwidth delay to R so that the packets leaving R are at least apart time T apart. But it will therefore keep packages of T time units apart, except for those times when the cwnd was just growing and A sends a couple of packets back in. Another package of such a turn back can be called an extra package. In order to simplify the argument somewhat, we will assume that two pair packages arrive in R essentially at the same time. Only an additional package can cause an increase in queue utilization; each other packet comes after the T interval from the previous package, giving R enough time to remove the packet from the queue. As a result, cwnd will reach the sum of transit and waiting dose capacity without R dumping the package. (This is not necessarily where such a large unit has been sent as one burst.) Let C be this combined capacity and assume that Cwnd has reached C. When A executes the next increase cwnd += 1 add-on, as usual, it will send a pair of back packets. The second of this steam – additional – has dried; will fall when it reaches the router bottleneck. At this point, C= cwnd − 1 packages are exceptional, all intervals in time intervals T. Sliding windows will continue normally until the ACK package shortly before the lost package comes back to A. After this point A will receive only dupACKs. A has received C=cwnd−1 ACKs since the last cwnd grow, but must receive C+1 = cwnd ACKs in order to increase cwnd again. This will not happen as the transfer of the lost package will no longer be the new AKK. After that, the cwnd decreases and the next sawdust tooth begins; the only packet that's lost is the extra package of the previous year. See 31.2.3 Single losses for trial validation and exercise 15.0. The TCP described above produced several embedded assumptions. Perhaps the most important thing is that any loss is seen as evidence of clogging. As we will see in the next chapter, this fails for TCP high bandwidth (when rare random losses become significant); also fails for TCP via wireless (either Wi-Fi or others), where lost packages are much more common than via Ethernet. See 21.6 Problem TCP High Bandwidth and 21.7 Problem With TCP's Lossy-Link. TCP's cwnd-increment strategy – to increase cwnd by 1 for each RTT - has some volume assumptions. This mechanism works well for between inland RTT in order of 100 ms, and for cwnd in the low hundreds. However, if cwnd = 2000, then 100 RTT - maybe 20 seconds - for cwnd growth should be 10%; lineano increase becomes relatively rather slow. Even if RTT is very long, the increase in cwnd is slow. The absolute minimum RTT light speed for geosynchronous satellite internet is 480 ms and the typical satellite-internet RTT closes to 1000 ms. moreover, as we will see below, such long RTT means that these TCP connections are poorly in competition with other connections. See 21.8 TCP satellite connection problem. Another implicit assumption is that if we have a lot of data to download, we will send everything in a single connection instead of splitting it between multiple links. Http's online protocol violates this routine, though. With several short links, cwnd may never be properly converged in a state of stability for any of them; TCP Reno does not support downloading what was learned about cwnd from one link to the next. A related problem occurs when the connection alternately changes between relatively uninterested periods and full data transmission; Most TCP sets cwnd=1 and returns to a slow start when sending CVs after a period of free time. Finally, TCP's rapid retransmit assumes that routers do not significantly redirect packets. In Reno's TCP Additive Increase, Multiplicative Decrease strategy, the increase in increase is 1.0 and the reduction factor is 1/2. It is natural to ask whether these values have any particular meaning or what the consequences are if they change. None of these values plays much of a role in determining the average cwnd value, at least in the short term; This largely imposes the capacity of the path, including the size of the router bottleneck waiting group. It seems clear that the exact value of the increase does not constitute a clogging; the increase in RTT is too small to have a major impact here. Reducing the factor 1/2 can play a role in the immediate response to auxiliary clogging by greatly reducing the cwnd at the first sign of lost packets. However, as we will see in 22.6 TCP Vegas, TCP Vegas in its usual mode works quite successfully with The strategy of reducing additives, reducing cwnd by 1 to the point where it detects the approximation of clogging (to make sure this detects well before the loss of the package), and by a few measures, responds better to clogging than TCP Reno. In other words, not only is the exact value of the AIMD reduction factor not critical to control clogging, but the multiplier reduction itself is not mandatory. There are two informal justifications in [JK88] to reduce the factor 1/2. The first is in a slow start: if Nth RTT finds that cwnd = 2N is too big, the sender falls back to cwnd/2 = 2N-1, which is known to have worked without loss of previous RTT. However, the change in the reduction policy can best be addressed by making a simultaneous change for a slow start; Alternatively, the 1/2 reduction factor is still used for a boundless slow start, while a new factor can β used for a marginal slow start. The second justification for the reduction factor 1/2 applies directly to the clogging avoidance phase; written in 1988, is quite remarkable for the modern reader: If the link is in a state of dynamic running and the package is off, it is probably because a new connection has started and took some of your bandwidth.... There are probably now two conversations that share bandwidth. So you should cut the window by half because the bandwidth available to you has been reduced by half. [JK88], §D Today, busy routers can have thousands of simultaneous connections. To be sure, Jacobson and Karels continue to state if there are more than two connections sharing bandwidth, halve their window is conservative – and being conservative at high traffic intensity is probably wise. This advice is still to this day. While they do not play a major role in determining CWND or avoiding congestive collapses, it turns out that these increases and reductions in 1 and 1/2 factors play a major role in fairness: ensuring that competing connections get the bandwidth allocation they should get. We will return to 20.3 TCP Reno Equity with synchronized losses and also 21.4 AIMD Revisited. The TCP Reno kernel clogging algorithm is based on algorithms in Jacobson and Charles's 1988 paper [JK88], which is now 50 years old, although NewReno and SACK are almost universally added to the standard Reno implementation. There are also broad changes in TCP usage patterns. Twenty years ago, the vast majority of all TCP traffic represented downloads from larger servers. Today, more than half of all Internet TCP traffic is peer-to-peer and not server-client. Increasing online video streaming creates new requirements for excellent TCP in real time. In the next chapter, we will examine the dynamic behaviour of TCP Reno, focusing in particular on fairness other problems of TCP Reno senders. We will then examine some attempts to solve these problems in 22 more recent TCP exercises. Tutorials are given fractional (floating points) numbers to allow interpolation of new exercises. Tutorials marked with ♢ solutions or hints on 34.14 TCP Reno solutions and Clogging Management. 1.0 Consider the next network, for any connection other than the first one, which has a bandwidth of 1 package/second. Assume that ACKS travel immediately from B to R (and thus to A). Assume that there are no delays in reproduction, so RTTnoLoad 4; bandwidth×RTT is then 4 packets. If A uses sliding windows with window size 6, the waiting bar on R1 will eventually have a size of 2. Let's say that A uses threshold slow start (19.2.2 Threshold Slow Start) with ssthresh = 6, and with cwnd initially 1. Complete the table below up to two rows for cwnd = 6; for these final two lines, cwnd has reached ssthresh and so A will send only one new package for each ACK received. Assume that the waiting order on R1 is large enough that the packets are not dumped. How big will the waiting order on R1 grow? T A sends R1 waiting times R1 sends B receptions/ACKs cwnd 0 1 1 1 1 2 3 4 2,3 3 2 2 5 5 3 2 6 7 8 4,5 5 4 2 3 Note, yes if, A, without the usual startao, A only after 6 package is off after 6 pack, a fine on R1 u start to hold 6-1 = 5 package. 2.0. Consider the next network from 19.2.3 Slow start multiple drops Example, with connections marked with bandwidth in packets/ms. Assume that ACKS travels immediately from B to R (and thus to A). A starts sending to B using an unlimited slow start starting with data[1] at T=0. Initially, cwnd = 1. Write a packet and delivery transfer table if the maximum queue size is R 4 (we do not count the packet that is currently being forwarded). Stop by coming to A first dupack, triggered by the arrival on the B of the package, which followed the first package, which was dumped by R. T A sends R waiting times to waiting times R sends B receptions/ACKs 0 Data[1] Data[1] 3.0. Consider the network from tutorial 2.0 above. Restarts mailing to B using a non-rems slow start, but this time the R is the size of the waiting group 2, without counting the package that is currently being forwarded. Make a table that shows all packet downloads with A, all batch drops by R, and other columns as useful. Assume that the retransmission mechanism is not used (no time cancellations, no rapid transmission), and that A only sends new data when it receives new ACK (dupACKs, in other words, does not initiate new data transfers). These assumptions will eventually stop new data transfers; continue the table until all portable data packets receive B. 4.0. The assumption that the link starts cwnd=1 and jumps cwnd for 1 each RTT without loss, and sets cwnd cwnd/2, rounded down on each RTT with at least one loss. Lost packets are not transmitted, and propagation delays prevail, so each window is sent more or less together. Packages 5, 13, 14, 23 and 30 are lost. What is the window size of each RTT, until the first 40 packages are sent? Which packages are sent to each RTT? Hint: Data is sent to the first RTT[1]. There is no loss, so in the second RTT cwnd = 2 and Data[2] and Data[3] is sent. 5.0. Supno supno that TCP Reno is used to transfer a large file over a path with a bandwidth high enough to be treated as a doubling of each RTT as in 19.2 Slow Start during slow start. Assume that the receiver does not limit the size of the window. (a). How many RTTs will it take for the window size to reach ~8,000 packages (about 213), assuming that there is an unmissable slow start and no packet loss? (b). About how many packages will be sent and validated by this point? (c). Assume that the bandwidth is 100 packets/ms and RTT is 80 ms, resulting in a bandwidth×delay of the product of 8,000 packets. What part of the total bandwidth will the link use to the point where the window size reaches 8000? Hint: The total bandwidth is 8,000 packets per RTT. 6.0. (a) Repeat the diagram in 19.4 TCP Reno and Fast Recovery, done there with cwnd=10, for window size 8. Assume that the data packet is lost[10]. There are seven dupACK[9]'s which can be conveniently identified as dupACK[9]/11 via dupACK[9]/17. When sending CVs, make sure you make it clear. (b). Let's say you try to do this with window size 6. Is this window size big enough for a speedy recovery to still work? If so, on which dupACK[9]/N begins a new data transfer? If not, what goes wrong? 7.0. We assume that the window size is 100, and data[1001] is lost. 99 dupACK[1000] will be sent, which can be transferred as dupACK[1000]/1002 via dupACK[1000]/1100. TCP Reno is used. (a). Where does dupACK[1000]/N sender start sending new information? (b). When the data transmitted[1001] arrive at the receiver, what is the ACK sent in response? (c). When the certificate referred to in (b) is returned to the sender, is the data packet sent? Tip: Efs term dupACK[1000]/N, for N≥1004. The third dupACK is dupACK[1000]/1004; what is the EFS at that time after the data retransmis[1001]? 8.0. Sup than say the window size is 40, and data[1001] is lost. Package 1000 will normally be pulled. Packages 1001-1040 will be sent and 1002-1040 will each be triggered by a duplicate ACK[1000]. (a). Which actual data packets trigger the first three dupACK? (The first ACK[1000] triggers Data[1000]; this is not considered a duplicate.) (b). After receiving the third dupACK[1000] and after receiving the lost data[1001], how much should the consignor assess the flight as such? When the data downloads[1001] arrives at the receiver, ACK[1040] will be sent back. (c). What is the first data [N] for which the answer is ACK[N] for N>1000? (d). What is the first N for which data[N+20] is sent in response to ACK[N] (this represents the point at which the link returns to normal sliding windows with window size 20)? 9.0. Recall (19.2 Slow start) to increase cwnd by 1 for each ACK received during slow start, resulting in the transmission of two new data packets. We assume that the slow start is modified by sending three new packages, not two, to each ACK; cwnd will now triple for each RTT, taking the value 1, 3, 9, 27, ... (a). For any future ACK, how much must now cwnd increase? (b). We assume that the pathway is mostly delayed by reproduction. Gradually larger window vessels are shipped, with the sizes of successive power 3, until cwnd is reached, where the package is lost. What window size can the sender reasonably be sure of, or is it working on the basis of previous experience? 10.0 For example, 19.5 TCP NewReno data[4] was not lost. (a) When data[1] is received, what would the ACK send in response? (b). At what point of the diagram can the sender continue the normal scrolling windows with cwnd = 6? 11.0. It is assumed that in the case of 19.5 TCP NewReno, Data[1] and Data[2] no data[4] were lost. (a). The third dupACK[0] is sent in response to data? (b). When the transmitted data[1] reaches the receiver, the ACK[1] answer is. When this ACK[1] reaches the sender, which data packets are sent in response? 12.0. Let's say that two TCP connections have the same RTT and share a bottleneck on which there is no other traffic. The size of the bottleneck waiting group is negligible compared to the bandwidth × RTTnoLoad. Loss events occur at regular intervals and are fully synchronized. Show that the two connections together will use 75% of the total narrow connection capacity, as in 19.7 TCP and Bottleneck Link Utilization (there's a single connection). See also practice 16.0 of Chapter 21 Further dynamics of TCP. 13.0. In 19.7 TCP and Bottleneck Link Utilization we showed that if the capacity of the waiting line was a router bottleneck of 50% transit capacity of TCP Reno connection, and there was no other traffic, then the utilization of the bottleneck-connection will be 95.8%. (a). We assume that the waiting capacity is 1/3 of transit capacity. Show the utilization of the bottleneck is 11/12 or 91.7%. Draw a tooth diagram and look for the relative lengths of unsaturated and waiting phases. You can round the cwndmax at 4/3 transit capacity (value cwnd shortly before batch loss; the exact value of the cwndmax is higher by 1). (b).♢ Formula to use a link in terms of f<1 waiter ratio transit capacity. Make the same simplification assumption as part (a). 14.0. In 19.2.1 TCP Reno Per-ACK Responses, we are tired of the TCP sender's response to increase the CWND as follows: (a). What is the appropriate formulation if the window size is actually measured in byteh and not in turns? Let SMSS mean the maximum size of the sender segment and let bwnd =SMSS×cwnd denote the congestion window as measured in bytes. Tip: Solve this last equation for cwnd and plug the result above. (b). What is the appropriate formulation cwnd = cwnd + 1/cwnd if delayed ACKs are used (18.8 TCP Delayed ACKS) and we still want the cwnd to increase by 1 for each window? Let's assume we're back to ing out cwnd in packages. 15.0. In 19.8 Single packet losses, we slightly simplified the claim by assuming that when sending A-packets to R they essentially came at the same time. Give a scenario in which there is no lost extra package (the other pair) but a package that follows it. Tip: See 31.3.4.1 Single sender phase effects. Effects.

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