1 Introduction

1.1 Architecture Priciples

1.1.1 Packets, Connections, and Datagrams

In **statistical multiplexing**, traffic is mixed together based on the arrival statistics or timing pattern of the traffic.

Alternative techniques, such as time-division multiplexing **(TDM)** and **static multiplexing**, typically reserve a certain amount of time or other resources for data on each connection.

Note that while circuits are straightforwardly implemented using TDM techniques, **virtual circuits (VCs)** that exhibit many of the behaviors of circuits but do not depend on physical circuit switches can be implemented atop connection-oriented packets.

The per-flow state is established prior to the exchange of data on a VC using a signaling protocol that supports connection establishment, clearing, and status information. Such networks are consequently called **connection-oriented**.

1.1.2 The End-to-End Argument and Fate Sharing

The function in question can completely and correctly be implemented only with the knowledge and help of the application standing at the end points of the communication system.

Fate sharing suggests placing all the necessary state to maintain an active communication association (e.g., virtual connection) at the same location with the communicating endpoints.

1.1.3 Error Control and Flow Control

1.2 Design and Implementation

12 TCP: The Transmisssion Control Protocol(Premilinary)

12.2 Introduction to TCP

12.2.1 The TCP Service Model

TCP provides a connection-oriented, reliable, byte stream service.

12.2.2 Reliability in TCP

If a segment arrives with an invalid checksum, TCP discards it without sending any acknowledgment for the discarded packet.

When TCP sends a group of segments, it normally sets a single retransmission timer, waiting for the other end to acknowledge reception. TCP does not set a different retransmission timer for every segment. Rather, it sets a timer when it sends a window of data and updates the timeout as ACKs arrive.

This acknowledgment may not be sent immediately but is normally delayed a fraction of a second. The ACKs used by TCP are cumulative in the sense that an ACK indicating byte number N implies that all bytes up to number N (but not including it) have already been received successfully.

This provides some robustness against ACK loss—if an ACK is lost, it is very likely that a subsequent ACK is sufficient to ACK the previous segments.

Once a connection is established, every TCP segment that contains data flowing in one direction of the connection also includes an ACK for segments flowing in the opposite direction.

12.3 Tcp Header and Encapsulation

```
0                             15 16                             31
-----------------------------------------------------------------
    Source Port(16)              |    Destination Port(16)
-----------------------------------------------------------------
                        Sequence Number(32)
-----------------------------------------------------------------
                        Acknowledgment Number(32)
-----------------------------------------------------------------
 Header  Reserv  C E U A P R S F|
 Length   (4)    W C R C S S Y I|       Window Size(16)
   (4)           R E G K H T N N|
-----------------------------------------------------------------
        Checksum(16)             |       Urgent Pointer(16)
-----------------------------------------------------------------
                              Options
kind(1) | lenth(1)   | info(n)
0                    |   EOL
1                    |   NOP
2           4        |   MSS
3           3        |   WSCALE/WSOPT
4           2        |   SACK
5           N*8 + 2  |   [,]...[,]
8           10       |   TSOPT: TSV(4) TSER(4)
-----------------------------------------------------------------
```

The combination of an IP address and a port number is sometimes called an **endpoint or socket** in the TCP literature.

The **Sequence Number** field identifies the byte in the stream of data from the sending TCP to the receiving TCP that the first byte of data in the containing segment represents.

The **ACK** Number contains the next sequence number that the sender of the acknowledgment expects to receive.

The sequence number of the first byte of data sent on this direction of the connection is the ISN plus 1 because the SYN bit field consumes one sequence number.

As we shall see later, consuming a sequence number also implies reliable delivery using retransmission. Thus, SYNs and application bytes (and FINs, which we will see later) are reliably delivered. ACKs, which do not consume sequence numbers, are not.

This **urgent pointer** is a positive offset that must be added to the Sequence Number field of the segment to yield the sequence number of the last byte of urgent data.

To be more efficient, multiple packets must be injected into the network before an ACK is received. This approach is more efficient but also more complex. A typical approach to managing the complexity is to use sliding windows, whereby packets are marked with sequence numbers, and the window size bounds the number of such packets. When the window size varies based on either feedback from the receiver or other signals (such as dropped packets), both flow control and congestion control can be achieved.

13 Connection Management

13.2 Tcp Connection Establishment and Termination

13.2.1 Tcp half-close

13.2.2 Simultaneous Open and Close

A simultaneous open requires the exchange of four segments, one more than the normal three-way handshake.

With a simultaneous close the same number of segments are exchanged as in the normal close. The only real difference is that the segment sequence is interleaved instead of sequential.

13.2.3 Initial Sequence Number

[RFC0793] specifies that the ISN should be viewed as a 32-bit counter that increments by 1 every 4μs. The purpose of doing this is to arrange for the sequence numbers for segments on one connection to not overlap with sequence numbers on a another (new) identical connection.

13.2.6 Connections and Translator

By implementing a portion of the TCP state machine in a NAT (see, for example, Sections 3.5.2.1 and 3.5.2.2 of [RFC6146]), the connection can be tracked, including the current states, sequence numbers in each direction, and corresponding ACK numbers. Such state tracking is typical for NAT implementations.

13.3 Tcp Options

Note that the MSS option is not a negotiation between one TCP and its peer; it is a limit. When one TCP gives its MSS option to the other, it is indicating its unwillingness to accept any segments larger than that size for the duration of the connection.

13.3.2 Selective Ackownledgment(SACK)

Once this has taken place, the TCP receiving out-of-sequence data may provide a SACK option that describes the out-of-sequence data to help its peer perform retransmissions more efficiently. SACK information contained in a SACK option consists of a range of sequence numbers representing data blocks the receiver has successfully received. Each range is called a SACK block and is represented by a pair of 32-bit sequence numbers. Thus, a SACK option containing n SACK blocks is (8n + 2) bytes long. Two bytes are used to hold the kind and length of the SACK option.

13.3.3 WSCALE/WSOPT

1. This option can appear only in a SYN segment.
2. The 1-byte shift count is between 0 and 14 (inclusive). A shift count of 0 indicates no scaling. The maximum scale value of 14 provides for a maximum window of 1,073,725,440 bytes (65,535 × 214), close to 1,073,741,823 (230 −1), effectively 1GB. TCP then maintains the "real" window size internally as a 32-bit value.
3. To enable window scaling, both ends must send the option in their SYN segments.

13.3.4 Timestamp Option and Protection Against Wrapped Sequence Number

The main reason for wishing to calculate a good estimate of the connection’s RTT is to set the retransmission timeout.

The PAWS algorithm does not require any form of time synchronization between the sender and the receiver. All the receiver needs is for the timestamp values to be monotonically increasing, and to increase by at least 1 per window of data.

13.3.5 User Timeout(UTO) Option

The UTO value (USER_TIMEOUT) specifies the amount of time a TCP sender is willing to wait for an ACK of outstanding data before concluding that the remote end has failed.

The UTO option allows one TCP to signal its USER_TIMEOUT value to its connection peer.

NAT devices could also interpret such information to help set their connection activity timers.

```
USER_TIMEOUT = min(U_LIMIT, max(ADV_UTO, REMOTE_UTO, L_LIMIT))
```

13.3.6 Authentication Option

13.4 Path MTU Discovery with TCP

13.5 Tcp State Transition

13.5.2 TIME_WAIT(2 MSL)

The final ACK is resent not because the TCP retransmits ACKs (they do not consume sequence numbers and are not retransmitted by TCP), but because the other side will retransmit its FIN (which does consume a sequence number).

Any delayed segments that arrive for a connection while it is in the 2MSL wait state are discarded.

tcp_tw_reuse(tcp_timestamps) **tcp_tw_recycle** **tcp_max_tw_buckets**

13.5.3 Quiet Time Concept

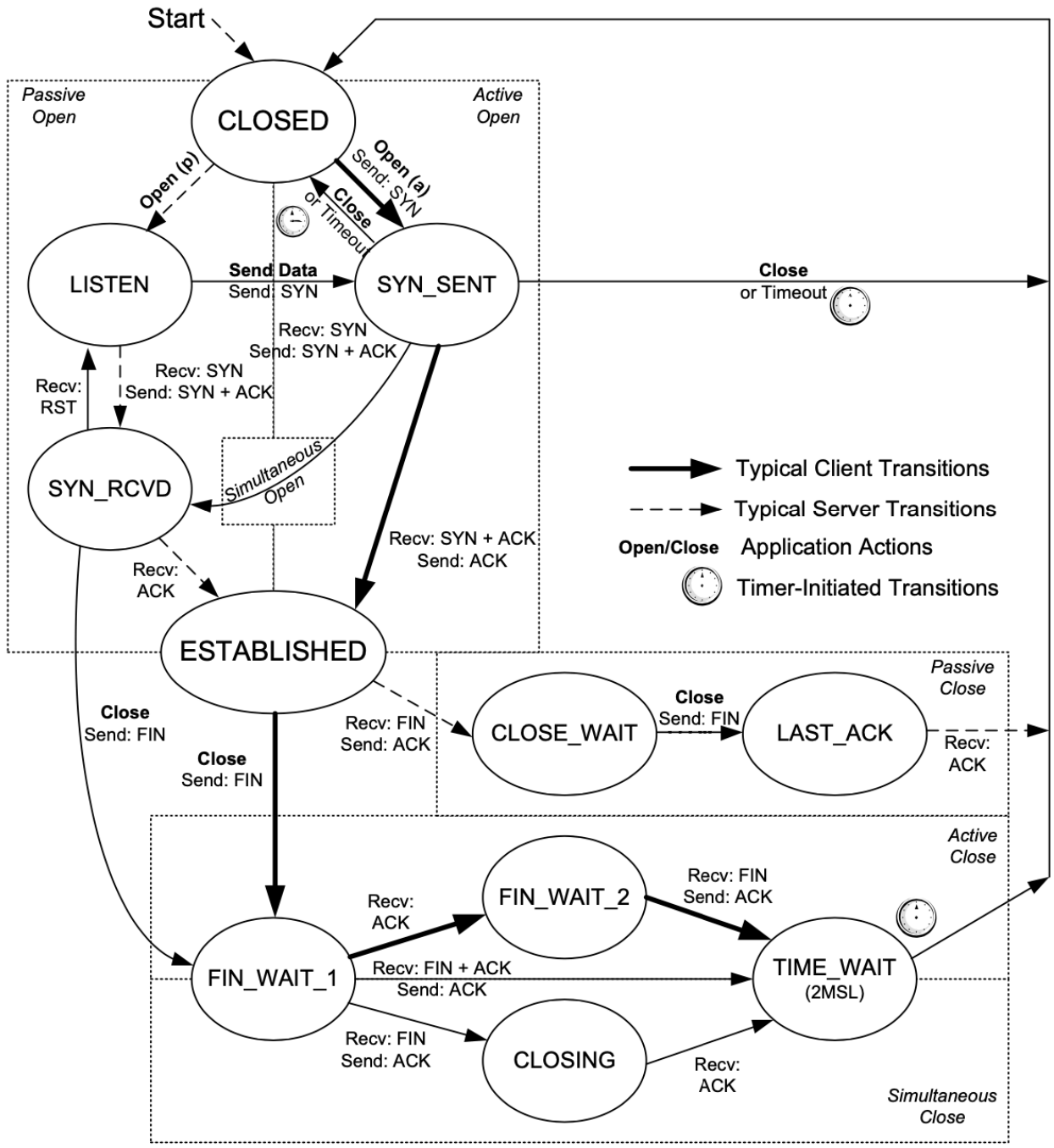
[RFC0793] states that TCP should wait an amount of time equal to the MSL before creating any new connections after a reboot or crash. This is called the **quiet time**. Few implementations abide by this because most hosts take longer than the MSL to reboot after a crash.

13.5.4 FIN_WAIT_2 State

One end of the connection can remain in this state forever. The other end is still in the CLOSE_WAIT state and can remain there forever, until the application decides to issue its close.

Solution: If the application that does the active close does a complete close, not a half-close indicating that it expects to receive data, a timer is set. If the connection is idle when the timer expires, TCP moves the connection into the CLOSED state.

13.5.5 Simultaneous Open and Close Transitions



Both ends go from ESTABLISHED to FIN_WAIT_1 when the application issues the close. This causes both FINs to be sent, and they probably pass each other somewhere in the network. When its peer's FIN arrives, each end transitions from FIN_WAIT_1 to the CLOSING state, and each endpoint sends its final ACK. Upon receiving a final ACK, each endpoint's state changes to TIME_WAIT, and the 2MSL wait is initiated.

13.6 RESET Segments

A reset is sent by TCP whenever a segment arrives that does not appear to be correct for the referenced connection.

Resets ordinarily result in a fast teardown of a TCP connection.

Circumstances:

1. connection to nonexistent port
2. aborting a connection
    - any queued data is thrown away and a reset segment is sent immediately
    - the receiver of the reset can tell that the other end did an abort instead of a normal close.
3. half-open close connections

    A TCP connection is said to be **half-open** if one end has closed or aborted the connection without the knowledge of the other end.
4. TIME-WAIT assassination
    - During TIME_WAIT state, the late-arriving segment is sent from the server to the client using sequence number L 100 and containing ACK number K 200. When the client receives this segment, it determines that both the sequence number and ACK values are "old." When receiving such old segments, TCP responds by sending an ACK with the most current sequence number and ACK values (K and L, respectively).
    - Most systems avoid this problem by simply not reacting to reset segments while in the TIME_WAIT state.

For a reset segment to be accepted by a TCP, the ACK bit field must be set and the ACK Number field must be within the valid window (see Chapter 12). This helps to prevent a simple attack in which anyone able to generate a reset.

13.7 TCP Server Operation

13.7.4 Incomming Connection Queue

Operating system ordinarily has two distinct connection queues:

1. Connections in SYN_RCVD state
2. Connections in ESTANBLISHED state but not acceped by server application

Linux rules on Connection Queue:

1. If the number of connections in the SYN_RCVD state would exceed **net.ipv4.tcp_max_syn_backlog**(default 1000) threshold, the incoming connection is rejected.
2. Berkeley listen API backlog value specifies only the maximum number of queued connections for one listening endpoint, all of which have already been accepted by TCP and are waiting to be accepted by the application.
3. If there is room on this listening endpoint's queue for this new connection, the TCP module ACKs the SYN and completes the connection. The server application with the listening endpoint does not see this new connection until the third segment of the three-way handshake is received. Also, the client may think the server is ready to receive data when the client's active open completes successfully, before the server application has been notified of the new connection. If this happens, the server's TCP just queues the incoming data.
4. If there is not enough room on the queue for the new connection, the TCP delays responding to the SYN, to give the application a chance to catch up.

13.8 Attacks Involing TCP Connection Management

SYN floods attack

> The main insight with SYN cookies is that most of the information that would be stored for a connection when a SYN arrives could be encoded inside the Sequence Number field supplied with the SYN + ACK.

SYN Cookies:

> The top 5 bits are (t mod 32) where t is a 32-bit counter that increases by 1 every 64s, the next 3 bits are an encoding of the server's MSS (one of eight possibilities), and the remaining 24 bits are a server-selected cryptographic hash of the connection 4-tuple and t value.

There are at least two pitfalls of this approach:

1. The scheme prohibits the use of arbitrary-size segments because of the encoding of the MSS.
2. The much less serious, connection establishment cycles that are very long (longer than 64s) do not work properly because the counter would wrap.

Degradation attacks with PMTUD

> an attacker fabricates an ICMP PTB message containing a very small MTU value (e.g., 68 bytes). This forces the victim TCP to attempt to fit its data into very small packets, greatly reducing its performance.

Solution:

1. disable PMTUD
2. disable PMTUD in cases where an ICMP PTB message with next-hop MTU under 576 bytes is received
3. insist that the minimum packet size (for large packets used by TCP) always be fixed at some value, and larger packets simply not have the IPv4 DF bit field turned on.

Hijacking

Spoofing attacks

> TCP segments that have been specially tailored by an attacker to disrupt or alter the behavior of an existing TCP connection.

Mitigation techniques:

1. authenticating each segment (e.g., using the TCP-AO option)
2. requiring reset segments to have one particular sequence number instead of one from a range
3. requiring particular values in the Timestamps option
4. using other forms of cookies in which otherwise noncritical data values are arranged to depend on more exact knowledge of the connection or a secret value.

14 TCP Timeout and Retransmission

14.1 Introduction

Fast retransmit is based on inferring losses by noticing when TCP's cumulative acknowledgment fails to advance in the ACKs received over time, or when ACKs carrying selective acknowledgment information (SACKs) indicate that out-of-order segments are present at the receiver.

TCP has two thresholds:

1. Threshold R1 indicates the number of tries TCP will make (or the amount of time it will wait) to resend a segment before passing "negative advice" to the IP layer (e.g., causing it to reevaluate the IP route it is using). **net.ipv4.tcp_retries1** default is 15(roughly 15~30minutes)
2. Threshold R2 (larger than R1) dictates the point at which TCP should abandon the connection. **net.ipv4.tcp_retries2** default is 3.

14.3 Setting the Retransmisstion Timeout(RTO)

TCP sends acknowledgments when it receives data, it is possible to send a byte with a particular sequence number and measure the time required to receive an acknowledgment that covers that sequence number.

The RTT is estimated for each TCP connection separately, and one retransmission timer is pending whenever any data is in flight that consumes a sequence number (including SYN and FIN segments).

14.3.1 The Classical Method

```
SRTT ← α(SRTT) + (1 − α) RTTs
RTO = min(ubound, max(lbound,(SRTT)β))
```

Problems with the classic method:

> the timer specified by [RFC0793] cannot keep up with wide fluctuations in the RTT (and in particular, it causes unnecessary retransmissions when the real RTT is much larger than expected).

14.3.2 The Standard Method

> - TODO

14.3.3 The Linux Method

It uses a clock granularity of 1ms, which is finer than that of many other implementations, along with the TSOPT.

> - TODO

14.3.4 RTT Estimator Behaviors

> - TODO

14.3.5 RTTM Robustness to Loss and Reordering

> - TODO

14.4 Timer-based Retransmission

When retransmission happens Tcp quickly reducing the rate at which it sends data into the network in two ways:

1. reduce its sending window size based on congestion control procedures.
2. keep increasing a multiplicative backoff factor applied to the RTO each time a retransmitted segment is again retransmitted.

14.5 Fast Retransmit

It is important to realize that TCP is required to generate an immediate acknowledgment (a "duplicate ACK") when an out-of-order segment is received, and that the loss of a segment implies out-of-order arrivals at the receiver when subsequent data arrives.

The duplicate ACKs sent immediately when out-of-order data arrives are not delayed.

Duplicate ACKs can also appear when there is packet reordering in the network—if a receiver receives a packet for a sequence number beyond the one it is expecting next, the expected packet could be either missing or merely delayed. Because we generally do not know which one, TCP waits for a small number of duplicate ACKs (called the duplicate ACK threshold or dupthresh) to be received before concluding that a packet has been lost and initiating a fast retransmit.

14.5.1 Example

The ACK at time 0.853 is not considered a duplicate ACK because it contains a window update.

The second retransmission is somewhat different from the first. When the first retransmission takes place, the sending TCP notes the highest sequence number it had sent just before it performed the retransmission (43401 + 1400 = 44801). This is called the **recovery point**.

TCP is considered to be **recovering from loss** after a retransmission until it receives an ACK that matches or exceeds the sequence number of the recovery point.

When **partial ACKs** arrive, the sending TCP immediately sends the segment that appears to be missing (26601 in this case) and continues this way until the recovery point is matched or exceeded by an arriving ACK. If permitted by congestion control procedures, it may also send new data it has not yet sent.

14.6 Retransmission with Selective Acknowledgments

A SACK-capable TCP receiver is able to describe data it has received with sequence numbers beyond the cumulative ACK Number field it sends in the primary portion of the TCP header.

A SACK option that specifies n blocks has a length of 8n + 2 bytes, so the 40 bytes available to hold TCP options can specify a maximum of four blocks. It is expected that SACK will often be used in conjunction with the TSOPT, which takes an additional 10 bytes (plus 2 bytes of padding), meaning that SACK is typically able to include only three blocks per ACK.

14.6.1 SACK Receiver Behavior

- A receiver generates SACKs whenever **there is any out-oforder data** in its buffer in two cases:
    1. data was lost in transit
    2. it has been reordered and newer data has arrived at the receiver before older data
- Behavior:
    1. The receiver places in the first SACK block the sequence number range contained in the segment it has **most recently received**.
    2. Other SACK blocks are listed in the order in which they appeared as first blocks in previous SACK options. That is, they are filled in by repeating the most recently sent SACK blocks (in other segments) that are not subsets of another block about to be placed in the option being constructed.

[RFC6675]Note that an ACK which carries new SACK data is counted as a duplicate acknowledgment under this definition even if it carries new data, changes the advertised window, or moves the cumulative acknowledgment point, which is different from the definition of duplicate acknowledgment in [RFC5681].

14.6.2 SACK Sender Behavior

The SACK sender keeps track of any cumulative ACK information it receives (like any TCP sender), plus any SACK information it receives.

It uses the SACK information it receives in ACKs generated at the receiver to avoid retransmitting data the receiver reports that it already has.

One way it can do this is to keep a "SACKed" indication for each segment in its retransmission buffer that is set whenever a corresponding range of sequence numbers arrives in a SACK.

When a SACK-capable sender has the opportunity to perform a retransmission, usually because it has received a SACK or seen multiple duplicate ACKs, it has the choice of whether it sends new data or retransmits old data.

The SACK information provides the sequence number ranges present at the receiver, so the sender can infer what segments likely need to be retransmitted to fill the receiver’s holes. The simplest approach is to have the sender first fill the holes at the receiver and then move on to send more new data [RFC3517] if the congestion control procedures allow. This is the most common approach.

14.6.3 Example

The ACK for 23801 contains a SACK block of [25201,26601], indicating a hole at the receiver. The receiver is missing the sequence number range [23801,25200], which corresponds to the single 1400-byte packet starting with sequence number 23801. Note that this SACK is a window update and is not counted as a duplicate ACK for the reasons discussed earlier. It does not trigger fast retransmit.

The SACK arriving at time 0.967 contains two SACK blocks: [28001,29401] and [25201,26601]. Recall that the first SACK blocks from previous SACKs are repeated in later positions in subsequent SACKs for robustness against ACK loss. This SACK is a duplicate ACK for sequence number 23801 and suggests that the receiver now requires two full-size segments starting with sequence numbers 23801 and 26601. The sender reacts immediately by initiating fast retransmit, but because of congestion control procedures, the sender sends only one retransmission, for segment 23801. With the arrival of two additional ACKs, the sender is permitted to send its second retransmission, for segment 26601.

14.7 Spurious Tiemouts and Retransmission

Caused by:

1. spurious timeouts (timeouts firing too early)
    - **Detection algorithm** attempts to determine whether a timeout or timer-based retransmission was spurious.
    - **Response algorithm** is invoked once a timeout or retransmission is deemed spurious. Its purpose is to undo or mitigate some action that is otherwise normally performed by TCP when a retransmission timer expires.
2. packet reordering, packet duplication
3. lost ACKs
4. when the real RTT has recently increased significantly beyond the RTO

14.7.1 Duplicate SACK(DSACK) Extension

With a non-SACK TCP, an ACK can indicate only the highest in-sequence segment back to the sender. With SACK, it can signal other (out-of-order) segments as well.

A receiver receives duplicate data segments can be the result of spurious retransmissions, duplication within the network, or other reasons.

D-SACK is a rule, applied at the SACK receiver and interoperable with conventional SACK senders, that causes the **first SACK** block to indicate the sequence numbers of a duplicate segment that has arrived at the receiver.

Purpose: to determine when a retransmission was not necessary and to learn additional facts about the network. With it, a sender has at least the possibility of inferring whether packet reordering, loss of ACKs, packet replication, and/or spurious retransmissions are taking place.

- Usage:
    1. A change is made to the content of SACKs sent from the receiver and a corresponding

change to the logic at the sender.
2. The change to the SACK receiver is to allow a SACK block to be included even if it covers sequence numbers below (or equal to) the cumulative ACK Number field.

DSACK information is included in only a single ACK, and such an ACK is called a DSACK. DSACK information is not repeated across multiple SACKs as conventional SACK information is. As a consequence, DSACKs are less robust to ACK loss than regular SACKs.

DSACKs, conversely, are able to be sent only after a duplicate segment has arrived at the receiver and able to be acted upon only after the DSACK is returned to the sender.

- How to identify a DSACK block within an ACK segment:
  1. if the first SACK block number < ACK number, it's DSACK
  2. else if the 2nd SACK number contains the 1st SACK number, the 1st SACK block is DSACK
  3. else it's normal SACK

Reference:

- https://www.cnblogs.com/lshs/p/6038617.html

14.7.2 The Eifel Detection Algorithem

The experimental Eifel Detection Algorithm [RFC3522] deals with this problem using the **TCP TSOPT** to detect spurious retransmissions.

After a retransmission timeout occurs, Eifel awaits the next acceptable ACK. If the next acceptable ACK indicates that the first copy of a retransmitted packet (called the original transmit) was the cause for the ACK, the retransmission is considered to be spurious.

The Eifel Detection Algorithm is able to detect spurious behavior earlier than the approach using only DSACK because it relies on ACKs generated as a result of packets arriving before loss recovery is initiated.

DSACKs, conversely, are able to be sent only after a duplicate segment has arrived at the receiver and able to be acted upon only after the DSACK is returned to the sender.

Mechanism:

- When a retransmission is sent (either a timer-based retransmission or a fast retransmit), the TSV value is stored. When the first acceptable ACK covering its sequence number is received, the incoming ACK's TSER is examined. If it is smaller than the stored value, the ACK corresponds to the original transmission of the packet and not the retransmission, implying that the retransmission must have been spurious.
- This approach is fairly robust to ACK loss as well. If an ACK is lost, any subsequent ACKs still have TSER values less than the stored TSV of the retransmitted segment. Thus, a retransmission can be deemed spurious as a result of any of the window's worth of ACKs arriving, so a loss of any single ACK is not likely to cause a problem.
- The Eifel Detection Algorithm can be combined with DSACKs. This can be beneficial in the situation where an entire window's worth of ACKs are lost but both the original transmit and retransmission have arrived at the receiver.

14.7.3 Foward-RTO(FRTO)

Forward-RTO Recovery (F-RTO) [RFC5682] is a standard algorithm for detecting spurious retransmissions. It does not require any TCP options, so when it is implemented in a sender. It attempts to detect only spurious retransmissions caused by expiration of the retransmission timer.

F-RTO makes a modification to the action TCP ordinarily takes after a timerbased retransmission. These retransmissions are for the smallest sequence number for which no ACK has yet been received. Ordinarily, TCP continues sending additional adjacent packets in order as additional ACKs arrive. This is the **go-back-N** behavior described previously.

- Mechanism:
    1. F-RTO modifies the ordinary behavior of TCP by having TCP send new (so far unsent) data after the timeout-based retransmission when the first ACK arrives. It then inspects the second arriving ACK.
    2. If either of the first two ACKs arriving after the retransmission was sent are duplicate ACKs, the retransmission is deemed OK
    3. If they are both acceptable ACKs that advance the sender's window, the retransmission is deemed to have been spurious.

This approach is fairly intuitive. If the transmission of new data results in the arrival of acceptable ACKs, the arrival of the new data is `moving the receiver's window forward`. If such data is only causing duplicate ACKs, there must be one or more holes at the receiver. In either case, the reception of new data at the receiver does not harm the overall data transfer performance (provided there are sufficient buffers at the receiver).

14.7.4 The Eifel Response Algorithem

14.8 Packet Reodering and Duplication

14.8.1 Reordering

- Contexts:
    - Packet reordering can occur in an IP network because IP provides no guarantee that relative ordering between packets is maintained during delivery. This can be beneficial (to IP at least), because IP can choose another path for traffic (e.g., that is faster) without having to worry about the consequences that doing so may cause traffic freshly injected into the network to pass ahead of older traffic, resulting in the order of packet arrivals at the receiver not matching the order of transmission at the sender.
    - some high-performance routers employ multiple parallel data paths within the hardware [BPS99], and different processing delays among packets can lead to a departure order that does not match the arrival order.
- Effect:
    1. If reordering takes place in the **reverse (ACK) direction**, it causes the sending TCP to receive some ACKs that move the window significantly forward followed by some evidently old redundant ACKs that are discarded. This can lead to an unwanted

burstiness (instantaneous high-speed sending) behavior in the sending pattern of TCP and also trouble in taking advantage of available network bandwidth, because of the behavior of TCP's congestion control (see Chapter 16).

2. If reordering occurs in the **forward direction**, TCP may have trouble distinguishing this condition from loss. Both **loss** and **reordering** result in the receiver receiving out-of-order packets that create holes between the next expected packet and the other packets received so far. When reordering is moderate (e.g., two adjacent packets switch order), the situation can be handled fairly quickly. When reorderings are more severe, TCP can be tricked into believing that data has been lost even though it has not. This can result in spurious retransmissions, primarily from the fast retransmit algorithm

A TCP receiver is supposed to immediately ACK any out-of-sequence data it receives in order to help induce fast retrans- mit to be triggered on packet loss, any packet that is reordered within the network causes a receiver to produce a duplicate ACK.

The problem of distinguishing loss from reordering is not trivial. Dealing with it involves trying to decide when a sender has waited long enough to try to fill apparent holes at the receiver. Fortunately, `severe reordering` on the Internet `is not common` [J03], so setting dupthresh to a relatively small number (such as the default of 3) handles most circumstances. That said, there are a number of research projects that modify TCP to handle more severe reordering [LLY07]. Some of these adjust dupthresh dynamically, as does the Linux TCP implementation.

14.8.2 Duplicaton

Although rare, the IP protocol may deliver a single packet more than one time. This can happen, for example, when a link-layer network protocol performs a retransmission and creates two copies of the same packet.

14.9 Destination Metrics

14.10 Repacketization

When TCP times out and retransmits, it does not have to retransmit the identical segment. Instead, TCP is allowed to perform repacketization, sending a bigger segment, which can increase performance.

15 Data Flow and Window Management

15.3 Delayed Ackowledgements

15.4 Nagle Algorithm

When a TCP connection has outstanding data that has not yet been acknowledged, small segments (those smaller than the SMSS) cannot be sent until all outstanding data is acknowledged.

Instead, small amounts of data are collected by TCP and sent in a single segment when an acknowledgment arrives.

This procedure effectively forces TCP into stop-and-wait behavior—it stops sending until an ACK is received for any outstanding data.

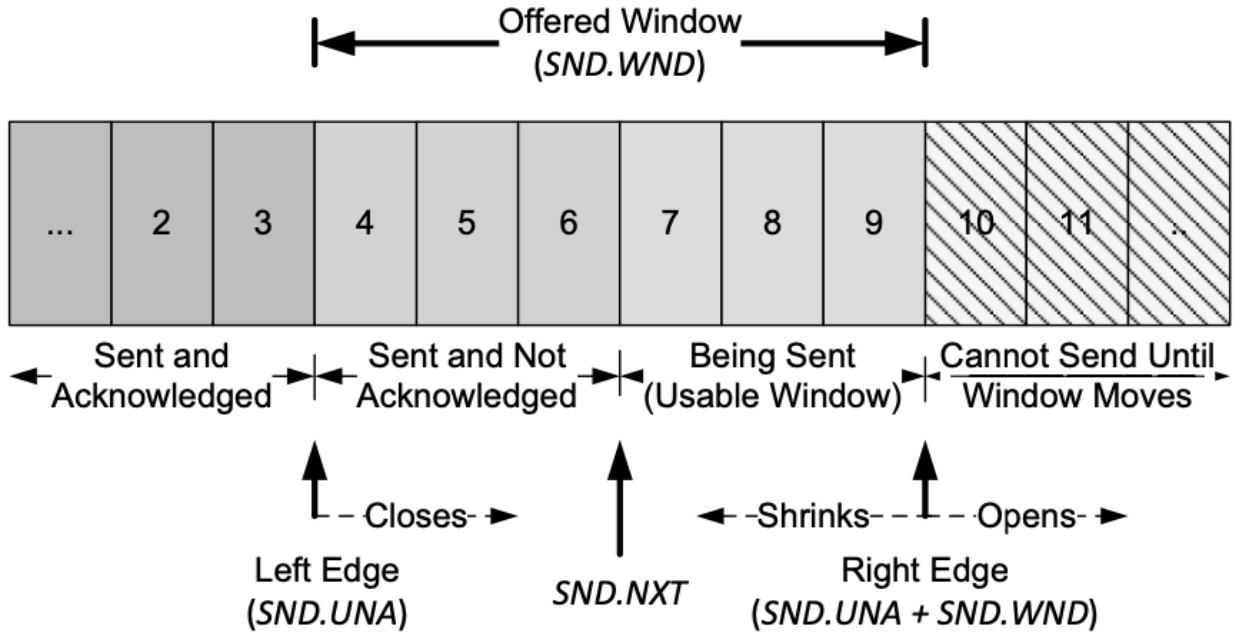
The trade-off the Nagle algorithm makes: fewer and larger packets are used, but the required delay is higher.

15.5 Flow Control and Window Management

The Window Size field in each TCP header indicates the amount of empty space, in bytes, remaining in the receive buffer. The field is 16 bits in TCP, but with the Window Scale option, values larger than 65,535 can be used.

15.5.1 Sliding Window

Sending Window

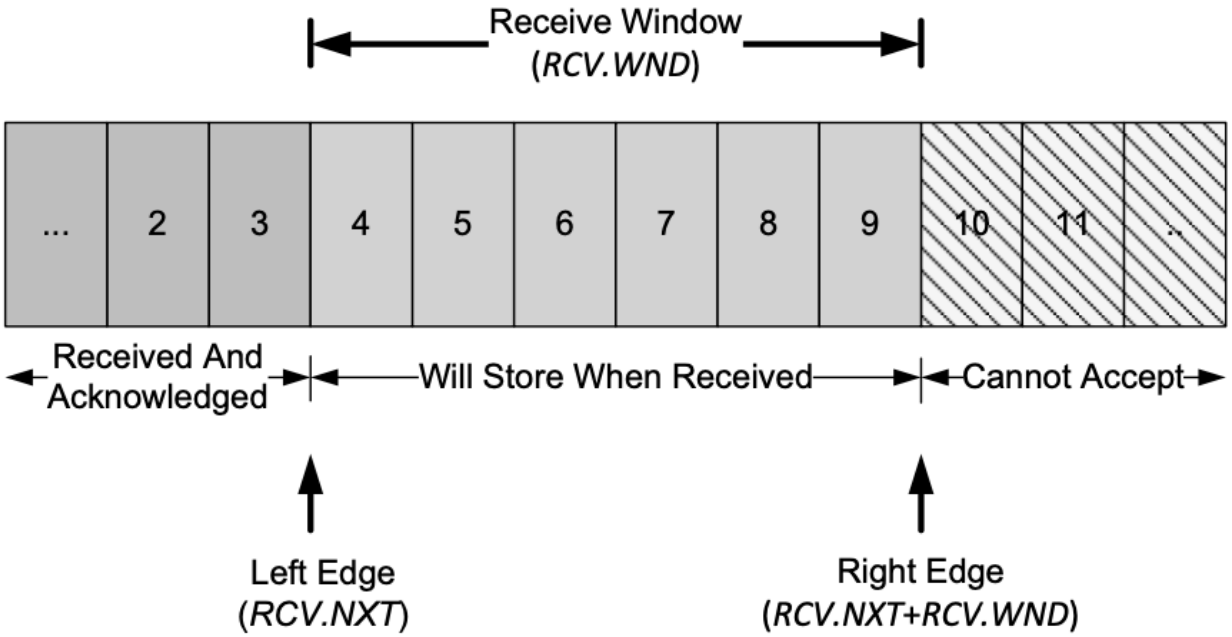


Sliding Process:

1. The window closes as the left edge advances to the right. This happens when data that has been sent is acknowledged and the window size gets smaller.
2. The window opens when the right edge moves to the right, allowing more data to be sent. This happens when the receiving process on the other end reads acknowledged data, freeing up space in its TCP receive buffer.
3. The window shrinks when the right edge moves to the left. The Host Requirements RFC [RFC1122] strongly discourages this, but TCP must be able to cope with it. Section 15.5.3 on silly window syndrome shows an example where one side would like to shrink the window by moving the right edge to the left but cannot.

TCP sender adjusts the window structure based on both values of **Ack number** and **window advertisement** whenever an incoming segment arrives

Receving Window



Note that the ACK number generated at the receiver may be advanced only when segments fill in directly at the left window edge because of TCP's cumulative ACK structure.

With selective ACKs, other in-window segments can be acknowledged using the TCP SACK option, but ultimately the ACK number itself is advanced only when data contiguous to the left window edge is received.

15.5.2 Zero Windows and TCP Pesist Timer

The sender uses a persist timer to query the receiver periodically, to find out if the window size has increased. The persist timer triggers the transmission of window probes. Window probes are segments that force the receiver to provide an ACK, which also necessarily contains a Window Size field.

The first probe should happen after one RTO and subsequent problems should occur at exponentially spaced intervals.

A normal TCP never gives up sending window probes, whereas it may eventually give up trying to perform retransmissions. This can lead to a certain resource exhaustion vulnerability

15.5.2.1 Example

There are numerous points that we can summarize using Figures 15-11 and 15-12:

1. The sender does not have to transmit a full window's worth of data.
2. A single segment from the receiver acknowledges data and slides the window to the right at the same time. This is because the window advertisement is relative to the ACK number in the same segment.
3. The size of the window can decrease, as shown by the series of ACKs in Figure 15-11, but the right edge of the window does not move left, so as to avoid window shrinkage.
4. The receiver does not have to wait for the window to fill before sending an ACK.

15.5.3 Silly Window Syndrome(SWS)

- Definition:
  - Small data segments are exchanged across the connection instead of full-size segments [RFC0813]. This leads to undesirable inefficiency because each segment has relatively high overhead—a small number of data bytes relative to the number of bytes in the headers.
- Solution:
  1. When operating as a receiver, small windows are not advertised. The receive algorithm specified by [RFC1122] is to not send a segment advertising a larger window than is currently being advertised (which can be 0) until the window can be increased by either **one full-size segment** (i.e., the receive MSS) or by **one-half of the receiver's buffer space**, whichever is smaller. Note that there are two cases where this rule can come into play: when buffer space has become available because of an application consuming data from the network, and when TCP must respond to a window probe.
  2. When sending, small segments are not sent and the Nagle algorithm governs when to send. Senders avoid SWS by not transmitting a segment unless at least one of the following conditions is true:
     - A full-size (send MSS bytes) segment can be sent.
     - TCP can send at least one-half of the maximum-size window that the other end has ever advertised on this connection.
     - TCP can send everything it has to send and either
       - an ACK is not currently expected (i.e., we have no outstanding unacknowledged data) or
       - the Nagle algorithm is disabled for this connection.

??? Figure 15-14, P749. Doese window probe segement break the normal data?

15.5.4 Large Buffers and Auto-Tuning

With auto-tuning, the amount of data that can be outstanding in the connection (its bandwidth-delay product, an important concept we discuss in Chapter 16) is continuously estimated, and the advertised window is arranged to always be at least this large (provided enough buffer space remains to do so). This has the advantage of allowing TCP to achieve its maximum available throughput rate (subject to the available network capacity) without having to allocate excessively large buffers at the sender or receiver ahead of time.

```
net.core.rmem_max = 131071
net.core.wmem_max = 131071
net.core.rmem_default = 110592
net.core.wmem_default = 110592

net.ipv4.tcp_rmem = 4096 87380 174760 // minimum, default, maximum
net.ipv4.tcp_wmem = 4096 16384 131072
```

15.6 Urgent Mechanism

// TODO

15.7 Attack Involving Window Management

// TODO

16 TCP Congestion Control

16.1.1 Dection of Congestion in TCP

In TCP, an assumption is made that a lost packet is an indicator of congestion, and that some response (i.e., slowing down in some way) is required.

16.1.2 Slowing Down a TCP Sender

```
W = min(cwnd, awnd) // awnd: actual quantity of data outstanding in the network
```

The TCP sender is not permitted to have more than W unacknowledged packets or bytes outstanding in the network.

The total amount of data a sender has introduced into the network for which it has not yet received an acknowledgment is sometimes called the **flight size**, which is always less than or equal to W.

When TCP does not make use of selective acknowledgment, the restriction on W means that the sender is not permitted to send a segment with a sequence number greater than the sum of the highest acknowledged sequence number and the value of W.

Bandwidth-delay Product This is the amount of data that can be stored in the network in transit to the receiver. It is equal to the product of the RTT and the capacity of the lowest capacity (“bottleneck”) link on the path from sender to receiver.

16.2 The Classic Algorithm

16.2.1 Slow Start

- Time:
    1. When a new TCP connection is created
    2. When a loss has been detected due to a retransmission timeout (RTO)
- Purpose:
    1. Help TCP find a value for cwnd before probing for more available bandwidth using congestion avoidance
    2. Establish the ACK clock.

[RFC5681]:

> Beginning transmission into a network with unknown conditions requires TCP to slowly probe the network to determine the available capacity, in order to avoid congesting the network with an inappropriately large burst of data. The slow start algorithm is used for this purpose at the beginning of a transfer, or after repairing loss detected by the retransmission timer.

```
cwnd = min(N, SMSS)

IW = 1 SMSS
IW = 2*(SMSS) and not more than 2 segments (if SMSS > 2190 bytes)
IW = 3*(SMSS) and not more than 3 segments (if 2190 ≥ SMSS > 1095 bytes)
IW = 4*(SMSS) and not more than 4 segments (otherwise)

cwnd = min(N, SMSS)
// N is the number of previously unacknowledged bytes ACKed by the received
"good ACK".
```

Some TCPs arrange to delay ACKs only after the connection has completed slow start. In Linux, this is called quick acknowledgments (**quickack mode**).

When Congestion happens, **cwnd** is reduced substantially (to half of its former value):

```
cwnd = cwnd / 2
```

16.2.2 Congestion Avoidance

```
cwndt+1 = cwndt + SMSS * SMSS/cwndt

cwnd0 = k*SMSS
cwnd1 = cwnd0 + (1/k)*SMSS
```

```
When receive an ack:    cwnd = cwnd + 1/cwnd
After an RTT:           cwnd = cwnd + 1
```

16.2.3 Selecting Between Slow Start and Congestion Control(When Congestion Happens)

The initial value of ssthresh may be set arbitrarily high (e.g., to awnd or higher), which causes TCP to always start with slow start. When a retransmission occurs(J: congestion happens), caused by either a retransmission timeout or the execution of fast retransmit, ssthresh is updated as follows:

```
ssthreh = cwnd / 2 | ssthresh = max(flight size/2, 2*SMSS)
cwnd = SMSS
```

Here we seen that if a retransmission is required, TCP assumes that the operating window must have been too large for the network to handle. Reducing the estimate of the optimal window size is accompanied by altering *ssthresh* to be about half of what the current window size is (but not ever below twice the SMSS).

16.2.4 Tahoe, Reno, and Fast Recovery

Tahoe

When congestion happens caused by either RTO or fast retransmit:

1. sshthresh = cwnd /2
2. cwnd = MSS
3. slow start unitl cwnd > ssthresh

Reno

One problem with Tahoe slow start is that it can cause the connection to `significantly underutilize` the available bandwidth while the sending TCP goes through slow start to get back to the point at which it was operating before the packet loss.

If packet loss is detected by duplicate ACKs (invoking fast retransmit), cwnd is instead reset to the last value of ssthresh instead of only 1 SMSS.

1. sshthresh = cwnd /2
2. cwnd = sshthresh + MSS
3. slow start unitl cwnd > ssthresh

Fast Recovery

Any ACKs that are received, even while recovering after a loss, still represent opportunities to inject new packets into the network. This became the basis of the **fast recovery** procedure. Fast recovery allows cwnd to (temporarily) grow by 1 SMSS for each ACK received while recovering.

Any nonduplicate (“good”) ACK causes TCP to exit recovery and reduces the congestion back to its pre-inflated value.

16.2.5 Standard TCP(Reno)

TCP begins a connection in slow start (cwnd = IW) with a large value of ssthresh, generally at least the value of awnd. Upon receiving a good ACK (one that acknowledges new data), TCP updates the value of cwnd as follows:

```
Slow start:             cwnd += SMSS (if cwnd < ssthresh)
Congestion avoidance:   cwnd += SMSS*SMSS/cwnd (if cwnd > ssthresh)
```

When fast retransmit is invoked because of receipt of a third duplicate ACK, the following actions are performed:

1. ssthresh is updated to no more than the value

```
ssthresh = cwnd / 2 | ssthresh = max(flight size/2, 2*SMSS)
```

2. The fast retransmit algorithm is performed, and cwnd is set to

```
cwnd = ssthresh + 3*SMSS
```

3. cwnd is temporarily increased by SMSS for each duplicate ACK received.
4. When a good ACK is received, set

```
cwnd = ssthresh (deflation) The actions in steps 2 and 3 constitute fast recovery.
```

Refer: https://coolshell.cn/articles/11609.html

16.3 Evolution of the Standard Algorithms

16.3.1 NewReno

One problem with fast recovery is that when multiple packets are dropped in a window of data, once one packet is recovered, a good ACK can be received at the sender that causes the **temporary window inflation** in fast recovery to be erased before all the packets that were lost have been retransmitted.

NewReno modifies fast recovery by keeping track of the highest sequence number from the last transmitted window of data (**recovery point**). Only when an ACK with an ACK number at least as large as the recovery point is received is the inflation of fast recovery removed.

This allows a TCP to continue sending one segment for each ACK it receives while recovering and reduces the occurrence of retransmission timeouts, especially when multiple packets are dropped in a single window of data.

16.3.2 TCP Congestion Control with SACK

In the case of fast retransmit/recovery, when a packet is lost, the sending TCP transmits only the segment it believes is lost and is able to send new data if the window W allows.

Using only cwnd as a bound on the sender's sliding window to indicate how many (and which) packets to send during recovery periods is not sufficient.

SACK TCP underscores the need to separate the congestion management from the selection and mechanism of packet retransmission.

One way to implement this decoupling is to have a TCP keep track of how much data it has injected into the network separately from the maintenance of the window. This is called **pipe variable**, an estimate of the flight size.

Pipe variable, an estimate of the flight size, counts bytes (or packets, depending on the implementation) of transmissions and retransmissions, provided they are not known to be lost.

A SACK TCP is permitted to send a segment anytime the following relationship holds true: cwnd pipe ≥ SMSS.

16.3.3 Forward Acknowledgment (FACK) and Rate Halving

When cwnd is reduced after a fast retransmit, ACKs for at least one-half of the current window's outstanding data must be received before the sending TCP is allowed to continue transmitting. This is an expected consequence of reducing the congestion window by half immediately when a loss is detected.
??? It causes the sending TCP to wait for about half of an RTT and then send any new data during the second half of the same RTT, a more bursty behavior than is really required.

In an effort to avoid the initial pause after loss but not violate the convention of emerging from recovery with a congestion window set to half of its size on entry, **forward acknowledgment (FACK)** was described in [MM96].

The basic operation of **Rate-Havling with Bounding Parameter(RHBP)** allows the TCP sender to send one packet for every two duplicate ACKs it receives during one RTT. This causes the recovering TCP to have sent the appropriate amount of data by the end of the recovery period, but it spaces or paces this data evenly, rather than bunching all the transmissions into the second half of the RTT period. Avoiding the bunching or burstiness is advantageous because bursts tend to persist across multiple RTTs, stressing router buffers more than required.

To keep an accurate estimate of the **flight size**, RHBP uses information from SACKs to determine the **FACK**: the highest sequence number known to have reached the receiver, plus 1. Taking the difference between the highest sequence number about to be sent by the sender (SND.NXT) and the FACK gives an estimate of the flight size, not including retransmissions.

```
awnd (flight size) = snd.nxt — snd.fack
```

With RHBP, a distinction is made between the adjustment interval (the period when cwnd is modified) and the repair interval (when some segments are retransmitted).
The adjustment interval is entered immediately upon a loss or congestion indicator.
The final value for cwnd when the interval completes is half of the correctly delivered portion of the window of data in the network at the time of detection.

The following expression allows the RHBP sender to transmit, if satisfied:

```
(snd.nxt — snd.fack + retran_data) < cwnd
```

Rate halving is one of several ways of pacing TCP's sending procedure to avoid or limit burstiness.

Triggering Recovery:

```
snd.fack - snd.una > 3 * MSS || dupacks == 3
```

16.3.4 Limited Transmit

It's a small modification to TCP designed to help it perform better when the usable window is small.

Recall from the experience with Reno TCP that when operating with a small window, there may not be enough packets in the network to trigger the fast retransmit/recovery algorithms when loss occurs, as these algorithms typically require three duplicate ACKs to be observed prior to initiation.

Note that rate halving is one form of limited transmit.

16.3.5 Congestion Window Validation (CWV)

The sender's current value of cwnd decays over a period of nonuse, and ssthresh maintains the "memory" of it prior to the initiation of the decay.

The **idle sender** has stopped producing data it wants to send into the network; ACKs for all the data it has sent so far have been received.

The **application-limited sender** does have more data to send but has been unable to for some reason. Because:

1. the sending computer is busy doing other tasks.
2. some mechanism or protocol layer below TCP is preventing data from being sent.

The CWV algorithm work as follows: Whenever a new packet is to be sent, the time since the last send event is measured to determine if it exceeds one RTO. If so:

1. ssthresh is modified but not reduced—it is set to max(ssthresh, (3/4)*cwnd).
2. cwnd is reduced by half for each RTT of idle time but is always at least 1 SMSS.

```
cwnd = cwnd / 2; // each RTT
```

For application-limited periods that are not idle, the following similar behavior is used:

1. The amount of window actually used is stored in W_used.
2. ssthresh is modified but not reduced—it is set to max(ssthresh, (3/4)*cwnd).
3. cwnd is set to the average of cwnd and W_used

```
cwnd = (cwnd + W_used) / 2;
```

16.4 Handling Spurious RTOs-the Eifel Response Algorithm

Such spurious retransmissions arise in a number of circumstances relating to changes in the underlying link layer (such as cellular handoff) or sudden onset of severe congestion contributing to a large increase in RTT.

The Eifel Response Algorithm is aimed at handling the retransmission timer and congestion control state after a retransmission timer has expired

It is initiated after the first timeout-based retransmission is sent.

Its purpose is to undo a change to ssthresh when a retransmission is deemed to be spurious.

In all cases, before ssthresh is modified as a result of the RTO, it is captured in a special variable as follows: pipe_prev = min(flight size, ssthresh). Once this has been accomplished, a dectecton algorithme is invoked to determine if the RTO is spurious, if it is, the following steps are executed when an ACK arrives after the retransmission:

1. If a received good ACK includes an ECN-Echo flag, stop (see Section 16.11).
2. cwnd = flight size + min(bytes_acked, IW) (assuming cwnd is measured in bytes).
3. ssthresh = pipe_prev.

16.5 An Extended Example

16.5.2 Sender Pause and Local Congestion (Event1)

Local congestion is one of several reasons the Linux TCP implementation may be placed in the **Congestion Window Reducing (CWR)** state [SK02]. It starts by setting ssthresh to cwnd/2, and by setting cwnd to min(cwnd, flight size + 1).

```
ssthresh = cwnd / 2;
cwnd = min(cwnd, flight size + 1);
```

In the CWR state, the sender reduces cwnd by one packet for every two ACKs received until cwnd reaches the new ssthresh or the CWR state is exited for some other reason such as a loss event.

16.7 Sharing Congestion State

16.8 TCP in High-speed Enviroments

In high-speed networks with large BDPs (e.g., WANs of 1Gb/s or more), conven- tional TCP may not perform well because its window increase algorithm (the congestion avoidance algorithm, in particular) takes a long time to grow the window large enough to saturate the network path.

16.8.1 HighSpeed TCP (HSTCP) and Limited Slow Start

The experimental HighSpeed TCP (HSTCP) specifications [RFC3649][RFC3742] propose to alter the standard TCP behavior when the congestion window is larger than a base value **Low_Window**, suggested to be 38 MSS-size segments.

16.8.2 Binary Increase Congestion Control (BIC and CUBIC)

16.9 Delay-Based Congestion Control

16.9.1 Vegas

16.9.2 FAST

16.9.3 TCP Westwood and Westwood+

16.9.4 Compound TCP

16.10 Buffer Bloat

16.11 Active Queue Management and ECN

16.12 Attacks Involving TCP Congestion Control